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Audio Amplifiers
Evolution of the Species
By J. Martins

Show Report
AXPONA 2022
Back to Listening in Person
By Oliver A. Masciarotte

Practical Test & Measurement
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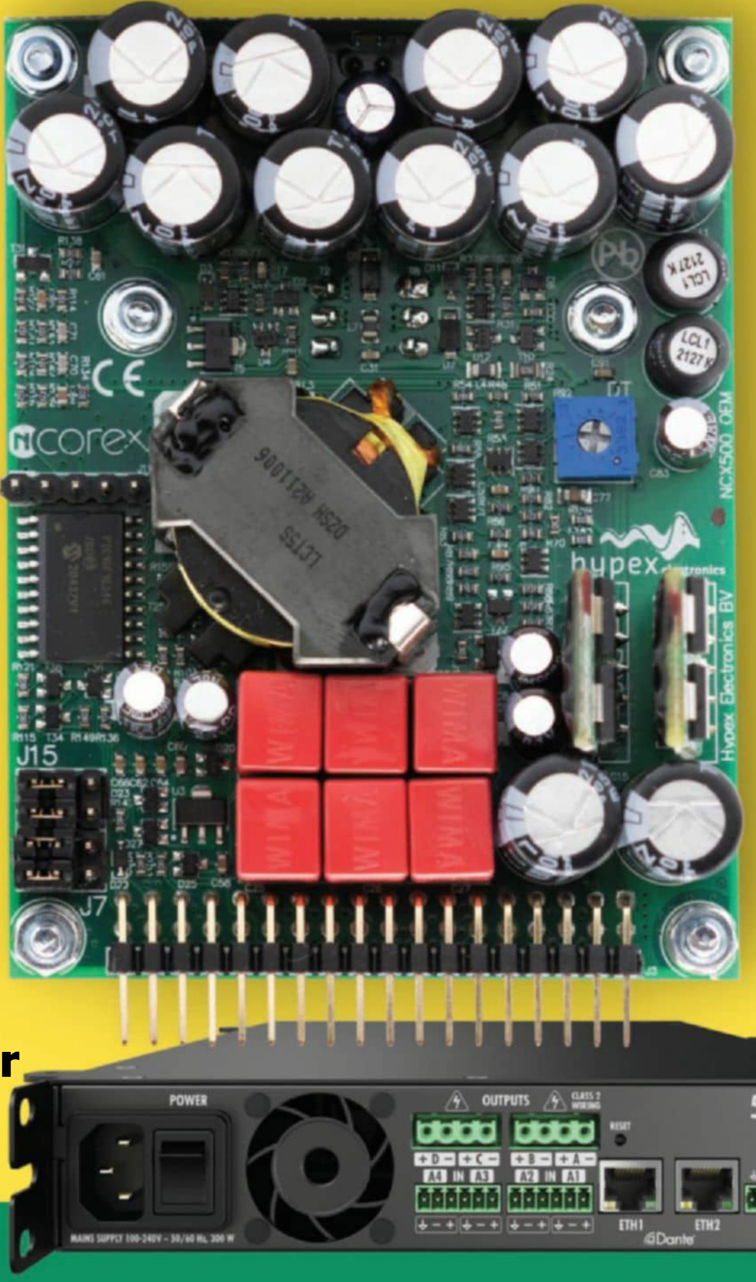
By Bruce Hofer

You Can DIY!
Egg Speaker Amp and System
By Reinhard Metz

The Cranberry Companion
A High-Performance Preamplifier Using Modular Construction
By Bruce Heran



Hollow-State Electronics
Triode, Pentode, or Beam Power Tube
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You Can DIY!
Building a Guitar-Controlled Synthesizer: Input Section & Pick Detector
By Ethan Winer

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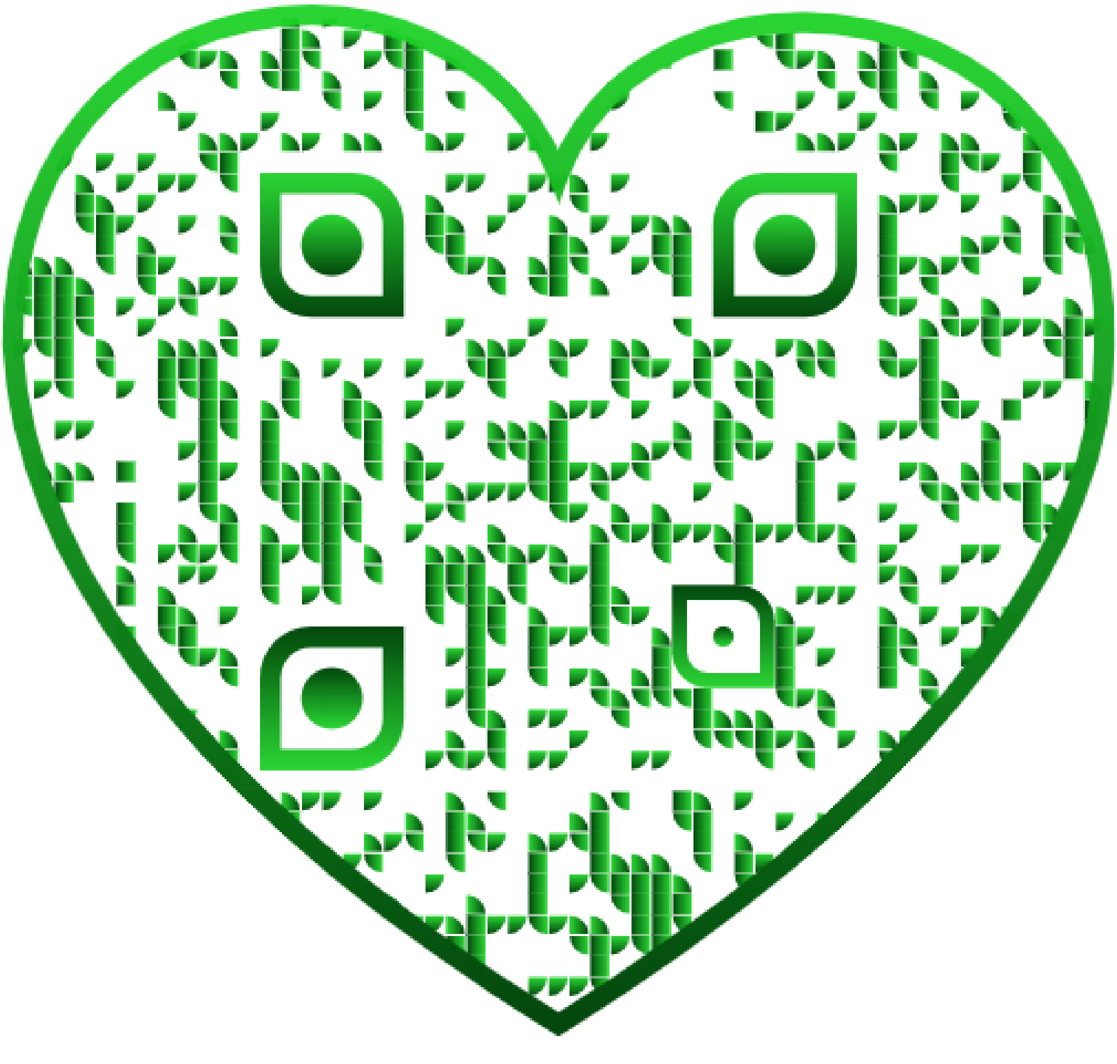

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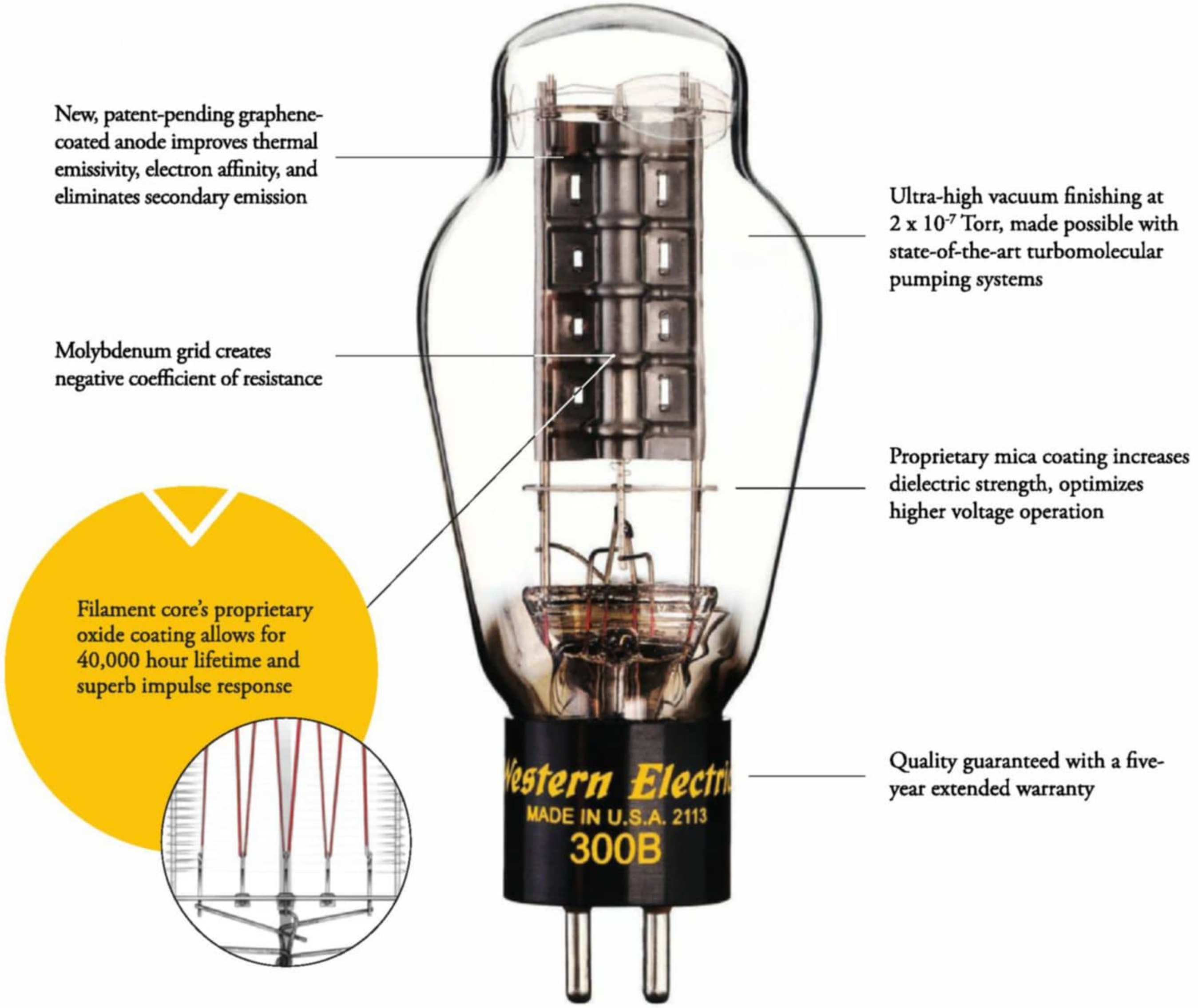

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P.O. Box 417
Chase City, VA 23924, US
Phone: 434-533-0246
Fax: 888-980-1303

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The Good, the Bad, and the Real Assessment

Over the past few weeks, we finally had the opportunity to attend the first trade shows since the start of the global pandemic, more than two years ago. And when I say trade shows, I mean real trade events with a significant number of exhibiting companies attending, and large numbers of visitors who were there to do business. Events with large numbers of new product unveils.

Whatever we say about our own—and our economy's—adaptability to a remote, online model, when people were confined at home and business needed to continue, nothing really replaces actual interaction at a trade show. Because it's not about the things we plan and need to do. It's mostly about the discovery, the unexpected, and everything that no one could have predicted.

And that's the main reason why many audio manufacturers have reserved some important announcements for an industry comeback, even though they could have planned a "virtual" web streaming presentation prior to shipping. But nothing replaces the physical experience, particularly when it comes to audio products—and that's good and bad.

In many cases, I find myself looking with wonder at products that I actually wrote about when they were announced. It was just that those announcements happened "virtually" and there's a disconnect with the actual experience of seeing, hearing, and touching. So, attending trade shows these days is like entering a candy store. But I also felt disappointment about some products that didn't impressed me, after touching or listening. Sometimes, revealing the effect of our emotional connections to a brand, a message, a design... and the disconnect with reality.

That should be obvious to someone aware of marketing techniques—but it's not that easy. It actually works exactly as detailed in scientific research about listening tests and how we are all conditioned by our own preconceptions, bias and autosuggestion.

As I had to focus on audio amplifiers for this issue of *audioXpress*, I couldn't help to question what manufacturers say about the performance of their products and technologies. Of course, there needs to be a bit of marketing, but reporting about multiple products that are only now available in prototype form or first production samples is extremely difficult to do objectively. That is why we were glad to at least receive some measurements from the actual manufacturers as was the case with the two recent announcements from Hypex Electronics of its new Nilai500 DIY amplifier module, and the new NCOREx Class D amplifier modules, the first of which is featured on our cover this month (together with the d&b audiotechnik 5D installation-focused amplifier and DPS with audio networking, also a remarkable new product).

While we wait for the opportunity to actually review these new amplifiers and perform our own set of measurements, we all share our readers' excitement about these technology evolutions and what they promise. That is why the trade show experience is invaluable, finally allowing the opportunity to actually listen and gather some real impressions—good and bad, largely subjective, but real.

This absence of contact with real products also should have taught us that we need to reinforce the way we make audio quality assessments based on those subjective listening impressions.

That is why I applaud the recent announcements of two new audio quality evaluation tools that will contribute to those much needed references. First is Multi-Dimensional Audio Quality Score (MDAQS), a new binaural perception-based software tool, intended to replace human evaluation with novel metrics in the assessment of audio quality. A much-needed approach for characterization of headphones, earphones, speakers, and any audio system, including automotive audio designs, researched, documented and validated by audio test and measurement company HEAD acoustics. Look for MDAQS on audioXpress.com.

The second related announcement was a new Preferred Listening Response Curve that could define the next standard in high-resolution earphones, the result from a new research by Knowles Corp. The research, intended to guide product designers, reflects the findings of an internal effort to determine the best and most satisfying music listening experience for consumers, and proposes an evolution of the Harman response curve specific to in-ears that takes into consideration frequencies above 10kHz. Again, search for Preferred Listening Response Curve on our website.

J. Martins
Editor-in-Chief

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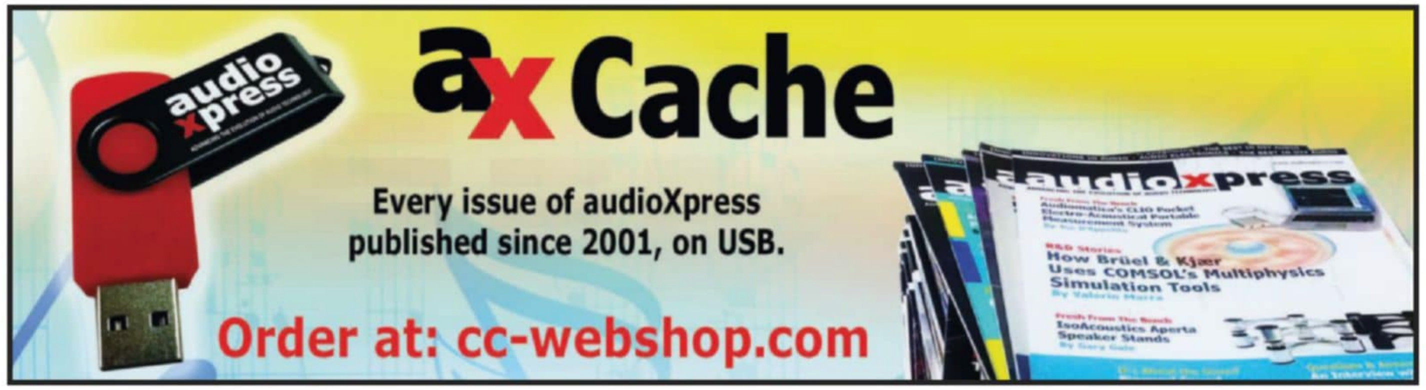

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AXPONA 2022

Back to Listening in Person

By
Oliver A. Masciarotte

More horns were in evidence at this year's show.

Score one for audiophiles. Our AXPONA expert found it a distinct pleasure to gather once again in the Chicago, IL, suburbs for the 2022 edition of the Audio Expo North America, better known as AXPONA. Everyone, exhibitors and attendees, had good fun while listening to some exceptional systems. In this show report, Oliver Masciarotte details some of the highlights.

Returning to the Renaissance Schaumburg Hotel & Convention Center for the 11th year, the show hosted close to 7,600 civilians. Masks were in evidence for those in need. I, having been jabbed multiple times, decided I needed my daily doses of wild SARS–CoV2 particles to complete my immunity. All good with that. The trouble was, I had a list of more than 50 vendors I specifically wanted to visit and then there were the 90 additional rooms that may have provided a moment of pleasurable audio serendipity. I had my work cut out for me…

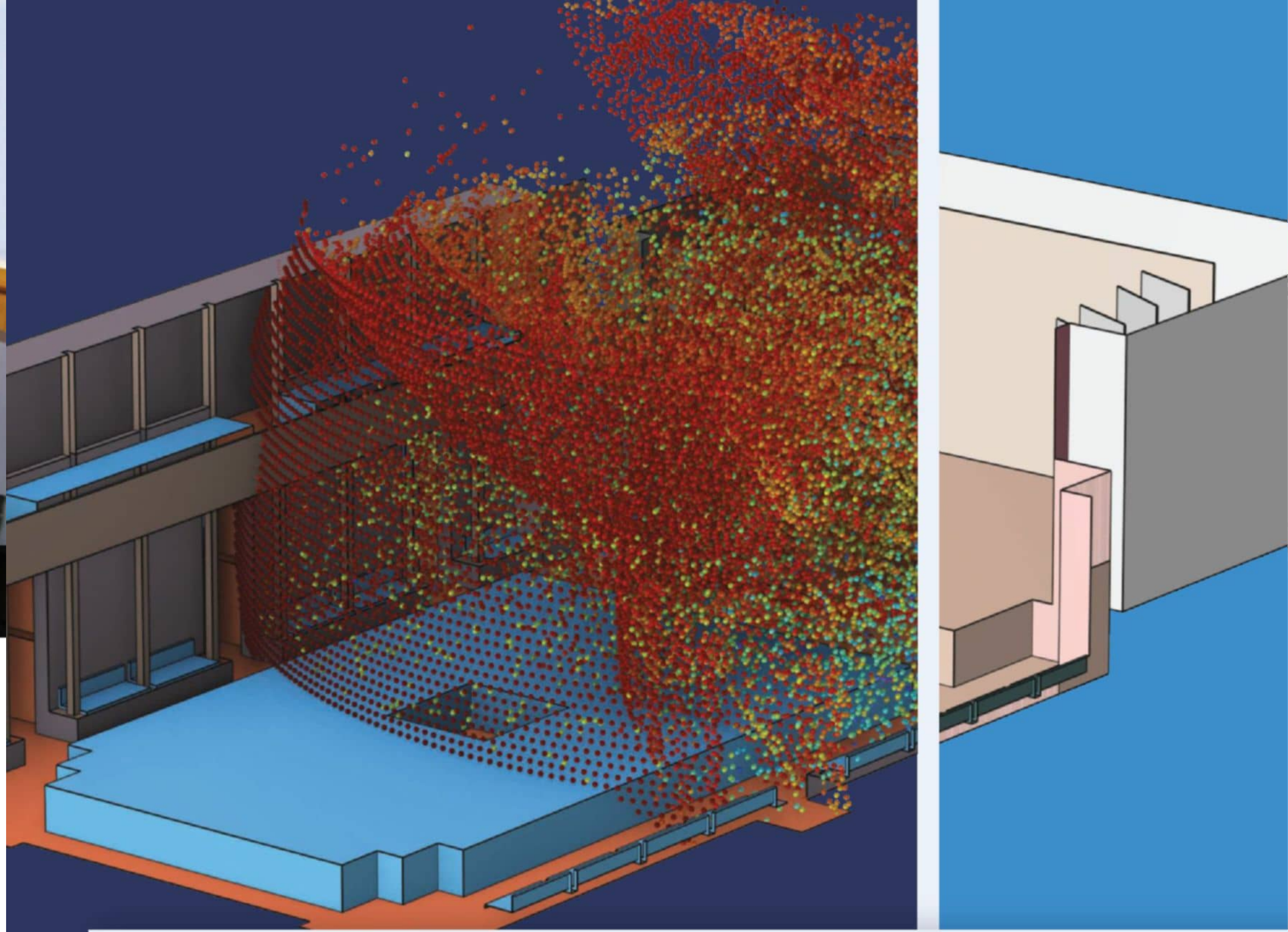
This Aries Cerat open baffle was folded to save space

Let's start with Zoltán Bay and his unique BAYZ radial transducer technology. Back in 2019, I heard the version 1 Counterpoint and was duly impressed. This year, it might have been the room but I found the smaller 2.0 version of the Courante to be less compelling. The imaging was, as my notes mentioned, amorphous while the low frequency was not as controlled and the overall timbre a bit on the sharp side.

An altogether different experience was had at Jim Salk's room, where his affordable and exceptional new BePure 2 was one of the top products for me at the show. This two-way floorstander combines SB Acoustic's Satori beryllium tweeter with dual 6.5" Purifi Audio midwoofers for a smooth, extended, and low distortion sound that's impossible to beat at their $6,000 price.

Tied for first place in the Real World Affordability category was Clayton Shaw's Spatial Audio $7,500 X4 Premium passive open-baffle speakers powered by LTA's $7600 Z40+ output transformerless integrated. Everything was wired up thanks to ANTICABLES. Both the Salk and Spatial rooms demonstrated how excellent a mid-priced system can sound.

I am not one to favor a skewed voicing choice when it comes to equipment. I like my speakers,



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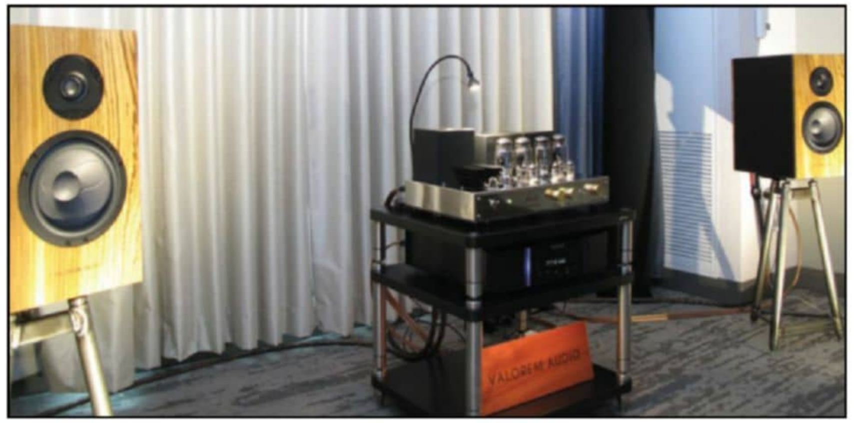
interconnects and electronics to be as linear and wide band as possible within the limits of budget. I feel that audio gear should be able to play acoustic chamber music just as well as EDM. That said, one speaker I heard would be an absolute blast to employ in a muscular, all-out rock-and-roll assault. The folks at Göbel High End had a COVID-new "small" loudspeaker that made me question my assumptions. The relatively squat Divin Marquis delivered "true high fidelity from subterranean bass to sparkling but not knife-edge highs," according to my notes. Göbel is known for BIG, and the Divin Marquis is no exception. At more than 3′ tall and 300 lb., even this "affordable" model comes in at $85,000.


At Valorem, hobby has become an obsession.

Back to the future with Augspurger's active MX-65

A Cabasse family portrait—the softball-sized PEARL KESHI satellite is shown at far right

Also showcasing a new speaker was local company Valorem Audio. Straight out of Addison, IL, designer and owner Dusan Plavsa started out as a hobbyist and DIY builder during quarantine. He later decided he wanted to get into it "…on a more permanent basis." For the show, Plavsa had his top of the line $8,400 Somnium stand mounters, powered by a Jadis 30W Class A attached with WireWorld Eclipse 8 cables. The Somniums rely on premium 1″ and 7″ Scan-Speak drivers plus audiophile crossover components in a vented, heavily braced Baltic birch ply cabinet. Various wood finishes come standard while real leather is also available as an option.

The Rare Audiovisuals

Audiovisuals are rare at mostly two-channel shows, but I have to mention Devialet and its sexy new soundbar. The diminutive $2400 Dione is dwarfed by my reference AMBEO, but what it lacks in size it makes up for in sound. Delivering a 5.1.2 experience, the bar accepts Airplay 2, Bluetooth 5, HDMI, and AES3 over F05 optical as inputs while acting as an UPnP end point.

As with other Atmos soundbars, the Dione doesn't fully envelope. Instead, it achieves a quarter sphere of rich and detailed sound. Unique to Dione is its centrally located ORB transducer, which can be physically rotated to change from tabletop to wall mounting.

What is not rare is the shuffling of personnel in the industry. Case in point; during the show it was announced that Andrew Jones, former propeller head at Elac, is now the Chief Loudspeaker Designer at MoFi. This means that not only will Jones most likely have freer rein to concentrate on just a handful of designs, but also MoFi will have a complete program for sales and marketing. From record labels all the way through to loudspeakers, they'll be able to offer a branded, end-to-end solution.

Pro Audio

A couple of pro audio vendors made the scene in the Ear Gear arena—RME and Rupert Neve Designs. The latter is no stranger to AXPONA, but Fort Lauderdale, FL-based Synthax chose the RME line from its very pro manufacturer collection (Ferrofish, Digigram, Appsys, etc.) to showcase. RME's versatile little ADI-2 DAC FS has so many bells and whistles, you simply have to visit www.synthax.com for all the details. Speaking of pro gear, the Lone Mountain group now represents the excellent Mutec line of clocking products, and were showing the REF10 universal format reference master clock and the MC3+ USB Smart Clock. While those were technically not new products, they are a new line for Lone Mountain to distribute. The $4999 REF10 exhibits ultra-low phase noise so that jitter, the Achilles heel of digital audio, can be reduced to vanishingly low levels. While the REF10 is aimed at pro setups, the $1649 MC3+ is specifically design to address the major shortcoming of USB DACs—their clocking. Of course, Lone Mountain was also filling its room with what I noted were beautiful sounds, thanks to one of my very favorite speaker brands, ATC. Some folks think soft domes are passé, but those folks in the UK have it dialed.

One more fabric dome aficionado is EgglestonWorks, which teamed up with Bryston to showcase its COVID-new (2021) $12,895 Oso three-way floorstander along with a new DAC/streamer/preamplifier. Bryston's BR-20 has got it all; digital ins via USB, AES3 on XLR along with S/PDIF on RCA and F05, and "fully discrete" unbalanced and balanced analog ins and Class A balanced outputs. They had an HDMI input option, but the company's supplier lamed out and it's no longer available. Including wired Ethernet for streaming and an optional MM phono stage, it ticks all the boxes at a reasonable $5,995 price point. Using the BR-20 as a streamer into a $6,795 $4B^3$ stereo power amp. Cabling was provided by Straight Wire, and power conditioning via Bryston's $4299 BIT20. Who knew that Bryston made power conditioners?! As to the speakers, EgglestonWorks' $12,900 Oso is a tallish drink of crystal clear water at 45″ and a gut busting 115 pounds each. Sporting a 1″ soft dome, dual 6″ mids and a side–firing 10″ woofer, they complemented a pleasantly real world-priced system. I am not necessarily a Bryston fanboy but this combo sounded really good.

Promising Newcomers

Hailing from Florida, Malbork Designs is a promising newcomer. Its minimalist $65,000 Warsaw combines a 4.7″ open-back AMT and dual 5″ aluminum-coned ferrite midrange drivers in an MTM arrangement. An SB Acoustics cast-aluminum basketed 8″ polypro woofer, also with a ferrite motor, completes the piece. The low-order hand-assembled crossover has the mids handing off at 5,500, while the woofer crosses at 350Hz. With the exception of the tweeter, each driver is housed in its own sealed space. In turn, the enclosures are mounted on vertically aligned, chiseled stalks. The entire speaker is fabricated from aluminum, and telegraphs an industrial, retro-future look. Speaking

Devialet's new high-performance Dione Atmos soundbar

Elliot Goldman and Göbel's Divin Marquis

About the Author

Oliver A. Masciarotte has spent more than 40 years immersed in the tech space, working on manufacturing, marketing, and product development for many pro and CE audio manufacturers including dbx, a/d/s, Lexicon, Sonic Solutions, and Minnetonka Audio. His client roster is as diverse as Apple, Harper Collins, NASA Johnson, NPR, and Universal. His writings include a book covering file-based music for the home and more than 100 articles for sundry trade publications. A member of the Audio Engineering Society (AES), Society of Motion Picture and Television Engineers (SMPTE), Project Management Institute (PMI), and Digital Cinema Society (DCS), he is currently co-founder and CMO of MAAT Inc., a digital audio software manufacturer.

The Renaissance Schaumburg Convention Center Hotel continues to be an ideal venue for the show.

A prototype omni sound sculpture from Malbork Designs

of which, Malbork also had a static display of a vertically-oriented, omni two-way desktop speaker. With fluidly sculpted acoustic lenses, the little beastie reminded me of a miniature version of Anthony Grimani lens design for Bang & Olufsen.

An additional manufacturer making its debut at AXPONA was Augspurger. Long a staple in the pro audio control room monitoring space, the distributor brought its littlest speaker system to the show. A hallmark of Augspurger speakers is their ability to cleanly deliver literally punishing sound pressure levels, hence their popularity with hip hop artists. Though I could not test that in their room, I'm confident that the new little MinimaX can play well at ear damaging volumes. That's not to say that the bookshelf-sized, active $10,995 two-ways can't also deliver at civilized sound pressure. Mid/highs for the quad front-ported model MX-65 are taken care of via a 1.4" beryllium diaphragm compression driver coupled to a compact maple horn, while low-frequency duties are handled by a long throw, 6.5" cone.

According to my notes, the rotatable, 70° × 110° multicellular horn provided very uniform, resonance-free coverage with a solid bottom. The included SXE-3D electronics package, incorporating balanced–only analog and 96k AES3 digital I/O, is housed in a 1U 19" rack–mountable case. I would love to be able to compare the MinimaX to Ocean Way's HR4 two-way mated to their S12A sub.

Next up, another "new" vendor at the show was Aretai, which had a $7,500 pair of Contra 100S standmounters on live demo. These good-looking little dudes are visually reminiscent of a Martion Einhorn, with a prominent, shallow flare horn-loaded SB ring tweeter mated to a front–firing 6" woofer and equally sized rear-mounted passive radiator. All that in a sealed enclosure containing a high-order crossover. Driven by already sterile Benchmark electronics, I found them to be mid-shy, but hey, it was a show so my 10-second listen could well have been skewed. I definitely want to understand Aretai better.

Overcoming Obstacles

Not deterred by a lost show shipment from their Austrian home, the Brodmann krew grabbed a local pair of entry-level $8,000 Festival F 2 from, if I remember correctly, a customer and ended up with an admirable showing. They were supposed to be exhibiting the far fancier $23,700 Vienna Classic VC 1. I found Brodmann's singular solution to low frequencies "interesting." I didn't get any details in time for publication but, according to my notes, they appear to include full-length side panels that provide sympathetic resonances.

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Impressive Showing

In the Alta/Mojo room, the new $15,000 three-way Adam floorstanders were special—mated to Mojo Audio's new $12,000 Mystique X DAC being fed from the also-new $4,998 Jay's Audio CDT3-MK3 Red Book disc transport with low noise/low jitter clock. Amplification was courtesy of Rogers High Fidelity $14,700 KWM-88 integrated. The whole exceeded the sum of those parts. Also refining the engineering was Gayle Sanders, with his new IMAGE.5 stand-mounted actives. As with his IMAGE1 floorstander, the soon-to-be-released half-pint will be driven by Böhmer Audio's wavelet integrated processor. I found this new entrant to provide a more believable low end than the larger version, with the active electronics weaving "…beautiful transient response into a more cohesive whole," according to my notes.

HiFi Rose goes "steampunk Nagra" with its sophisticated $7,000 RA180 GaN Class D integrated.

Salk Sound's estimable BePure 2 towers

Personal Audio Curiosity

In the personal audio department, I finally got to hear two products I've been curious about—Warwick Acoustics' new BRAVURA and Audeze's CRBN. Derived from Audeze's work on a non-magnetic headphone for medical imaging applications, the $4,500 "Carbon" electrostats employ a polymer/carbon nanotube alloy in place of vacuum-deposited metals. I found my too-brief listen to be "ethereal." Also electrostatic but on a different tack, Warwick's BRAVURA is designed to work with its existing M1 DAC/energizer/HPA. At a package price of $5,995 for cans + electronics, I still found Warwick's combination of voicing, transient response and overall presentation to be the best headphone experience around.

The Unexpected

As was the case at AXPONA 2019, Magnepan had an unadvertised location at the show where it was demoing its updated Little Ribbon Speaker (LRS). As in the past, the company was using a prototype amplifier driven by unknown source components, so I passed on the invitation until I could hear them with a reasonably predictable electronics chain. Let's hope they've made significant improvements.

Too Much to Hear in Too Little Time

In retrospect, I missed visiting several rooms while being lured into other spaces by the shear weirdness I glimpsed while passing. In particular, I had wanted to hear what Polk and Totem were up to, as I've followed those brands since CES was held at the Alexis Park. I also missed KEF's new $9,000 stand-mounted Reference 1 Meta since I have such a positive impression of its downmarket LS50 Meta.

In addition to KEF, here at *audioXpress* we've been following PerListen Audio and I lost an opportunity to get some seat time with them. According to CTO Erik Wiederholtz, they also felt they had some difficulty gaining attention for their new model during the gathering. "The S7t is our flagship, it's what was playing almost the whole time. The S5t was having its debut but everyone wanted to hear the S7t due to the accolades, so we mostly played that." The new S5t comes in at $6,995 to $7,995 each, depending on the finish, while the S7t is $8,995 to $9,995 each also depending on finish.

Bigger Is Not Always Better

In the Bigger-Is-Not-Necessarily-Better category, I happened upon a spendy system employing giant, open-baffle $137,500 Aries Cerat Aurora cubic loudspeakers from Cyprus powered by Pink Fawn and in-akustik electronics and cabling. Robert Neill, owner of distributor Worldwide Wholesales, walked me through the setup: in-akustik $4,300 AC-4004 2-meter cable supplying a $5,500 4500P conditioner, also by in-akustik. A $30,000 Pink Fawn 2.16 Ultra server/streamer sourced a $47,000 Aries Cerat Kassandra Ref DAC into the new $76,000 Aries Cerat Ageto Gen3 triode preamp, which in turn supplied signal to $50,000 Aries Cerat Essentia monoblocks. The Aurora loudspeakers have a couple of novel features, including their mini-biradial horn-loaded tweeter and Bluetooth-enabled crossovers for configuration on a mobile device. The sound in the room did not really capture my fancy but it sure was a visually impressive array of components.

Overall Impressions

For me, AXPONA 2022 was all about the return of in-person events and the rise of quality European vendors here on American shores. It's good to see more manufacturers are braving our domestic marketplace. Another trend to note with alarm was that our industry has gotten a bit smaller in recent history, with Exogal and several old skool hi-fi dealerships folding their tents during the COVID pandemic. The past few years have been tough on us all, and I can only hope our world will advance, not decline into chaos.

EIKON's curvy and kewl little IMAGE.5 are exceptional.

Evolution of the Species

Better, Smarter, Integrated, Connected, and Multichannel

In this Market Update we recap the most interesting events and announcements in the field of audio amplification, a key product and technology segment for the audio industry overall.

By
J. Martins
(Editor-in-Chief)

The last 12 months have been an exciting time for audio electronics, and in particular for audio amplification. Benefiting from expanded time for research and development activities caused by the global COVID-19 pandemic, companies in general have used the pause in many market activities to reinforce development. Of course, many other companies have also used the time to redefine their strategies, and we've seen some interesting company moves, market repositioning, and strategic alignments.

In this article, we review the most interesting announcements from the companies involved in audio amplification. But of course, to do that, we need to characterize the market conditions where everything is taking place. And front and center, we need to take into account the sequential and substantial market disruptions that originated in 2020 with the pandemic global expansion, continued in 2021, when it expanded in more serious ways to many regions that had not been previously affected. That sequence of events is still being felt in 2022, as some countries start to feel some relief from the pandemic, while others are now imposing even more severe pandemic control and confinement measures. This sequence inevitably caused an escalation on supply chain constraints and logistical problems at a global scale, which is now conditioning every industry and international trade. Add the invasion of Ukraine, Russian sanctions and market isolation, rising costs of fuel and energy with inevitable inflation, and anything that we assume going forward becomes unpredictable.

Anyway, that is not how companies can be managed, and audio companies, like all other organizations, work with road maps that need to be navigated, and strategic paths that need to be steered to move forward.

Naturally, predicting the behavior of a specific product segment such as audio amplification becomes a futile exercise under these circumstances. For perspective, I still think it is interesting to look at some numbers published just prior to these escalation of events.

And the first thing to note is that, when we discuss audio amplification today, we are talking about 96% of the overall business depending on Class-D audio amplifier technology. Yes, the other topologies are still important in product volumes, such as hi-fi or automotive.

As an example, the Class-G topology—which is essentially a modification of Class-AB amplifiers using rail switching to increase efficiency and reduce power dissipation—is still relevant in many designs, and used in automotive audio. And Class D/G combination amplifiers are currently meeting many of the needs of smart and wireless speakers in terms of quiescent power consumption. But in terms of the revenue this represents compared to overall sales of audio amplifiers, Class D has become the dominant topology for audio amplification.

That is well reflected in existing research. Insight Partners reported in November 2021, that the Class-D audio amplifier component market was valued at

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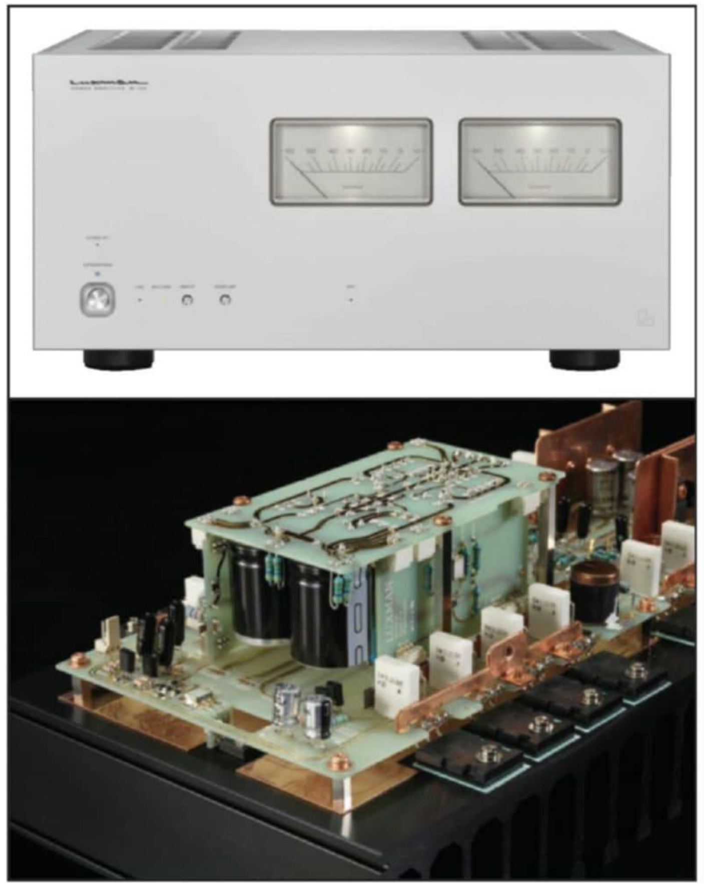
$2.4 billion USD in 2021 and is estimated to double in revenue in 2028, with an estimated compounded growth of 9.2% annually. The big boost is represented by the higher volume categories, starting with smart amplifiers for mobile devices, including personal audio devices, and automotive audio. Other major categories for Class-D audio amplifiers are televisions and home audio. This study focuses on audio component ICs exclusively. As The Insight Partners report highlights, AI-enabled smart speakers, consumer electronics, and appliances are a significant trend driving the proliferation of the Class-D audio amplifier market.

Curiously, the market research available completely ignores the segmentation by integration, which is currently a major trend, as we will describe in this report.



In the best tradition of Japanese classic hi-fi components, Luxman recently announced the Luxman M-10X power amplifier ($19,995 USD), its new stereo power amplifier that introduces the new Luxman Integrated Feedback Engine System (LIFES1.0) circuit. The new refined audio design is said to achieve the ideal balance between low distortion and effortless musicality using three-stage Darlington equipped quadruple paralleled push-pull output modules. To create this refined topology, Luxman engineers embarked on a systematic exploration of alternate designs, using simulation software, and selected FETs with unusually high transconductance (gm) and a different polarity in the primary stage, combined with a highly regulated constant voltage circuit in the drive stage, among many other technical refinements. The result is prodigious power into just about any conceivable load: not only 150W + 150W into 8Ω, but also an instantaneous 1,200W + 1,200W into 1Ω, or even 2,400W into 2Ω in BTL monaural mode.

Audio Amplifier ICs

Analog Devices (particularly since the merger with Maxim Integrated), Infineon, Cirrus Logic, On Semi, STMicroelectronics, Texas Instruments, Silicon Labs, and Qualcomm are all major players in the audio IC level market.

As an example, STMicroelectronics (ST), a company that leads in the automotive space with its FDA901 Class-D audio amplifier IC, launched in 2020 in cooperation with Japanese manufacturer Alps Alpine, in 2022 introduced its latest TDA7901 single-chip quad-bridge automotive amplifier using the company's advanced BCD (Bipolar-CMOS-DMOS) technology that integrates a buck controller for Class-G power switching, and supports high-resolution audio. The TDA7901 features a full D/A converter, digital input for direct connection to I²S (or TDM) and powerful MOSFET output stages, creating a unique proposition for high efficiency.

Overall, The Insight Partners research indicates that, together, the larger category of audio IC and audio amplifier devices represented more than $20 billion USD in 2020, and is projected to grow at an annual compounded rate of 7.7% to 2028. Key players considered on this wider study, apart from the semiconductor companies listed previously, include Yamaha, Realtek, Asahi-Kasei, ESS Technology, ROHM, Knowles, BSE, and others, considering the growing trend toward device integration and system integration with A/D and D/A conversion, DSP, interface, sensors, and more. This clearly indicates that we need to consider any market size figures very cautiously and understand how they have segmented the market.

Another relevant example, in 2019, Maxim Integrated Products—now part of Analog Devices—launched the MAX98390 smart amplifier with its advanced integrated Dynamic Speaker Management (DSM) algorithm that efficiently delivers louder, clearer sound at the lowest quiescent power. The boosted, digital Class-D DSM smart amplifier safely drives higher power levels (up to 5.1W) into tiny speakers typically rated for much lower power between 1W to 3W. This meets the miniaturization of consumer devices requirements, where microspeakers need to fit within smaller form factors, but loudness or sound pressure level (SPL) levels need to be maintained at the perceived levels of (almost) conventional speaker systems. Maxim solved the challenge by utilizing integrated IV (current and voltage) sensing, combined with its DSM algorithm to drive speakers to their maximum specified limits, while protecting against over excursion and over-temperature events. Effectively, the MAX98390 delivers up to 2.5× loudness (SPL) and up to 2 octaves deeper bass versus conventional 5V amplifiers in a small form factor.

Already in 2021, Maxim unveiled the MAX98396 Class D/G speaker amplifier, which the company claimed to have the industry's lowest noise and lowest quiescent power consumption—nearly 50% lower power than similar amplifiers. In this case, this met different requirements, since the MAX98396 amplifier features an inaudible noise floor, enabling placement in near-silent environments and offers 12.7mW of quiescent power to meet power compliance regulations for always-on portable Bluetooth and smart speakers.

The MAX98396 offers 12.7mW of quiescent power consumption at 12V Power Vdd (PVDD), which meets and exceeds the industry's standby power compliance regulations for always-on speakers. And this power efficiency allows the speakers to be driven higher and longer before reaching the thermal limit. With 45kHz of bandwidth and a dedicated data path, the MAX98396 offers an upgraded sampling frequency of up to 192kHz and an ultrasound signal bypass path to prevent signal attenuation by the audio processing. This feature, combined with artificial intelligence (AI) software and ultrasound, enables applications with touchless gesture sensing, presence detection and proximity sensing. Almost a new class of audio amplifier devices.

And while we are mentioning audio amplifier ICs that are currently being designed for low power, highly efficient designs, such as portable speakers, since its acquisition of Danish startup Merus Audio, Infineon Technologies has considerably expanded its portfolio of MERUS Class-D audio amplifier ICs, multi-chip modules, and discrete MOSFET/HEMT driver ICs. With its MERUS family of audio ICs using multi-level amplifier technology, which leads the industry in terms of power efficiency, Infineon offers a fully integrated solution that allows filter-less heatsink-free implementations, allowing audio systems optimization for performance, size, and cost.

Infineon's integrated Class-D audio amplifier multi-chip module (MCM) family of devices integrates PWM controller and digital audio power MOSFETs in a single package, eliminating the need for heatsinks, and shrinking PCB size by up to 70%.

The latest example from Infineon, already announced in 2022, is the new MERUS two-channel, analog input, Class-D audio amplifier MCM MA5332MS, a higher power and smaller footprint device added to the MERUS technology family. It is a powerful upgrade to its predecessor, offering the same or even higher output power than monolithic alternatives, without a heatsink and 50% less footprint.

Available in a compact 42-pin QFN package, the new MERUS MA5332MS integrates a dual-channel PWM controller, a high-voltage gate driver, and four low RDS(on) MOSFETs. Thanks to its very low RDS(on) Class-D output stage (24.4mΩ typical), it can deliver 2× 100W at 4Ω operating absolutely heatsink-free or 2× 200W at 4Ω with a small 8°C/W heatsink. Compared to other single-chip solutions, this is a considerably reduced size.

In the 100-400W/channel range, the MA5332MS is ideal for consumer products, including all active speakers, soundbars, subwoofers, and mini-component systems, all the way to professional audio applications in active studio monitors, guitar amplifiers, aftermarket car, and marine audio amplifiers.

Audio developers will also appreciate the fact that the MA5332MS offers differential or single-ended input and multiple output configuration options, single-ended (2× SE), bridge-tied load (BTL), and parallel single-ended (PSE), using a single or split power supply. Enabling an SE topology as an alternative to BTL allows for reduced bus capacitors and output low-pass filters.

As a highly-integrated MCM, the MA5332MS also integrates protection circuitry for functions such as over-current, over-temperature, and under-voltage protection with self-reset capability. This on-chip protection circuitry saves the time to design protection using external protection components.

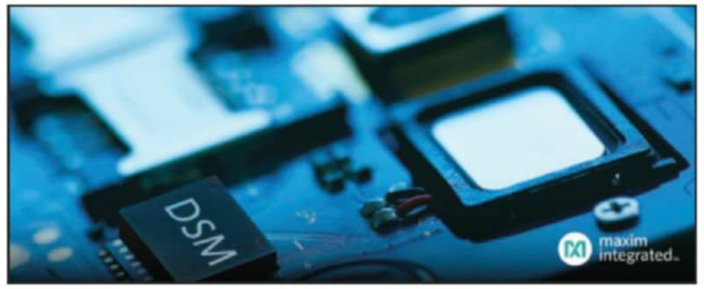
Amplifier Modules

While audio IC solutions and power electronics are increasingly the core focus for any audio product developer, we also need to address the audio amplifier module market, which is what most audio manufacturers in the home and professional audio markets mostly integrate in their system designs. That is again a very specialized segment, generating significant revenue within the product category, since unit sales prices are much higher, considering each module delivers an almost complete solution, including matching power supplies. This is a market that is currently dominated at the entry level by Asian-based companies, which even combine every single conceivable audio and network interface, wireless streaming and even audio sources, if required.

A possible example is Taiwan/China-based Arylic, which offers a continuously updated range of affordable DIY boards and plate amplifiers, all with wireless audio streaming support. A recent example is the Up2Stream Amp V4 board, a 2× 50W stereo amplifier

NAD Electronics launched a new and improved NAD Masters M10 V2 BluOS Streaming Amplifier, which continues its "just add speakers" lifestyle hi-fi philosophy. The M10 V2 features 100W per channel of nCore amplification, a 32-bit/384kHz ESS Sabre DAC, and BluOS streaming capabilities, adding the capability to expand with Dolby Digital Surround decoding, an enclosed IR remote control, and updated gain algorithms.

In Class-G operation, the TDA7901 buck controller from STMicroelectronics, automatically optimizes the voltage supplied to the bridge-tied load (BTL) power stage depending on the audio-signal level. The resulting analog signal comes with near-Class-D efficiency at normal listening levels. Integrating the buck controller in the IC contributes to reducing system size and weight, and also simplifies circuit design. The TDA7901 delivers 4× 43W of audio power (at 4Ω load, saturated output at 14.4V).



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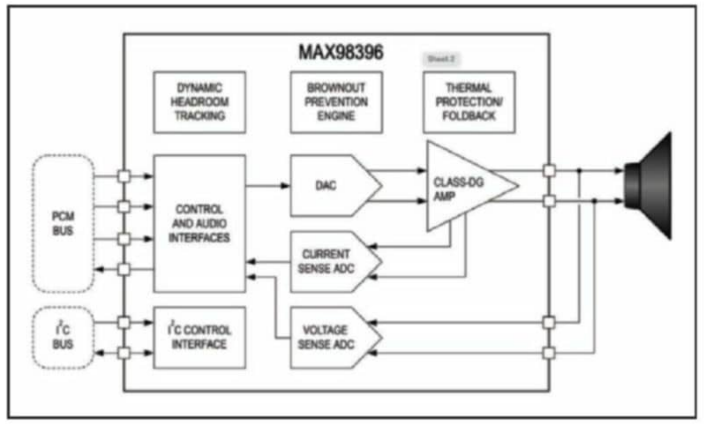
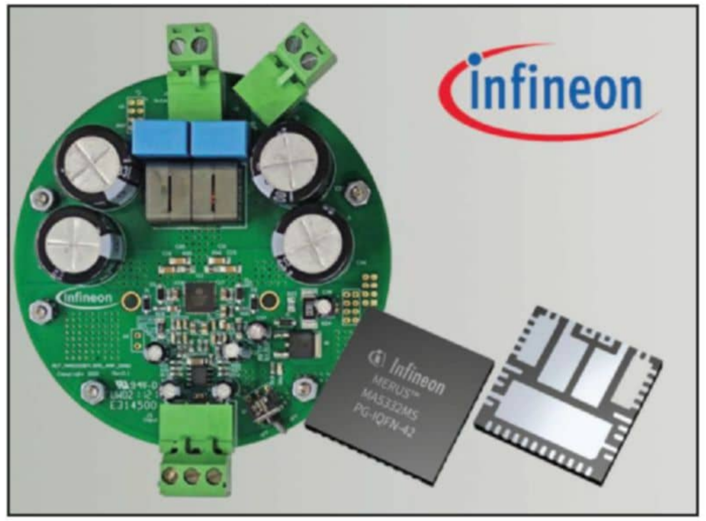


The Maxim Integrated MAX98396 Class D/G high-resolution audio amplifier was designed to meet the needs of smart and wireless speakers.



The new Infineon MERUS MA5332MS Class-D audio amplifier multichip module reduces cost, size, and thermal management concerns, providing more space for the industrial design and delivering superior efficiency and low output distortion. Also shown is the REF_MA5332BTLSPS reference board, a single BTL channel, 200W/channel (4Ω at 40V) Class-D audio power amplifier for home audio systems.

board that accepts 12V to 24V power inputs and 4Ω to 8Ω speakers. The board features WiFi, Airplay, Spotify Connect and Bluetooth 5 wireless capabilities, ideal for active wireless speakers, home theaters, and multiroom systems.

There are of course countless other audio amplifier boards, which in general leverage the audio technology and circuit designs that are generally available, using the before mentioned audio ICs. Most of the sales volumes in that segment is actually not for the DIY market, but for the cheap-and-convenient range of audio products that is available globally, and does particularly well in online sales.

For more demanding home audio quality, DIY or finished products, usually the audio amplifier design or technology becomes a strong sales argument. We will focus on the Class D efforts for this report, since that is where all the action is—and 2021/2022 is proving to be an exciting time on that front.

First, there's Hypex Electronics, the Dutch OEM specialist supplier in power amplifiers, power supplies and DSP solutions, which continues to lead in the high-end audio, hi-fi, and DIY market. Hypex has been cautious in separating its OEM business from the DIY market, creating the dedicated DIYclassD brand and website, which allows that important segment to buy specific products and kits, and get much needed support, without interfering with the activities of manufacturers who rely on Hypex as a technology supplier and close partner.

That is actually an approach that we would like to see implemented by other companies, and which unfortunately has not been well understood. In 2021, Hypex announced a new UcD700 mono kit featuring its UcD technologies marketed under its DIYclassD brand, making available the Hypex UcD700HG with HxR amplifier module, which has been around for many years but was never offered as a DIY building kit before. Now, DIYclassD offers an easy to build 700W monoblock kit including all the necessary parts that users don't even need to solder.

Hypex was founded in 1996 by Jan-Peter van Amerongen (1964-2021) as a supplier of plate amplifiers for live sound loudspeakers. The products drew the attention of hi-fi speaker manufacturers, resulting first in a line of active subwoofer amplifier subassemblies, shortly followed by multichannel units with active crossover filters for the studio market. In 2003, the company transitioned to the newly invented "Universal Class D" (UcD) technology, licensed from Philips, and later followed by NCore technology, after hiring UcD's inventor, Bruno Putzeys, to be the company's chief of R&D.

Now under the direction of Jan-Willem Winters, who in 2021 transitioned from managing the R&D department and has been with the company for nearly 12 years to becoming its CEO, Hypex is expanding its efforts both in the DIY and the OEM front.

Unveiled at this year's High End Show in Munich, Hypex launched the Nilai500DIY module, which is specified to deliver 500W with 1% THD at 4Ω and will improve output noise and idle losses significantly, compared to the popular NCORE NC400 module, which was already one of the most highly regarded (and best-selling products among audio enthusiasts). The Nilai500DIY module is a completely new amplifier module that sets a new benchmark, and was designed specifically for high-end audio applications, as the company highlights. In fact, Nilai is a completely new Class-D technology and a completely new amplifier module. As Hypex described, the amplifier takes Hypex control-loop topology to a whole new level resulting in an immense performance upgrade compared to the original Ncore series.

According to the company's R&D team, the massive performance boost—verified in the first production samples of the new Nilai500DIY module—results in distortion figures "that are 10 times lower, a power supply noise rejection that is 10 times higher and a reduced output impedance that has dropped by a factor of 10. These figures may already sound incredible, wait until you hear them come alive through your speakers!"

The new Nilai500DIY module comes with a discrete buffer stage and voltage regulators, just like the NC400 it will be replacing. And Hypex designed a new PS500DIY power supply to match, which includes a PFC and standby power supply. According to the first measurements supplied to audioXpress (and available online in an exclusive article), the frequency response within the audible range is exceptional. The Nilai500DIY features an input gain that can be adjusted with jumpers in three different settings, allowing to achieve the lowest output noise of 9.5μV with the lowest gain setting of 11.8dB. This is including the buffer stage and everything else. Not just the power stage. The reported THD+N at 100W in 4Ω is 0.00015% from 20Hz to 20kHz, which comes down to a signal-to-noise ratio (SNR) of -116dB. With idle losses of only 2.2W compared to 4.5W for the existing NC400 module, Hypex also managed to get the heat dissipation much lower. Again, this is including buffer stage. The output impedance of the Nilai500DIY is <50 μΩ at 1kHz.

And specifically for the OEM market, Hypex also announced this year a new and improved iteration of its popular NCORE Class-D amplifier technology and unveiled the NCx500 OEM, the first new NCOREx amplifier module to be launched. According to the Dutch company, the new technology will afford at least 2× better performance compared to original NCORE products. This is of course big news for many manufacturers in the audio industry who have long relied on Hypex amplifier modules and more specifically on NCORE products.

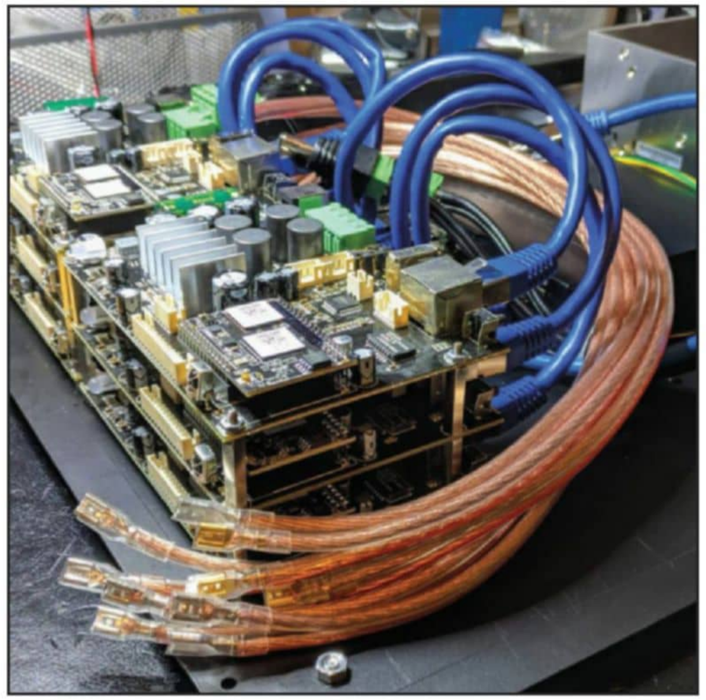
The first-ever module that implements the new NCOREx technology is the NCx500 OEM, which can be used as a drop-in replacement for the existing NC500 OEM module. The new NCx500 module features an on-board, high-end buffer stage combining Hypex' established HxR discrete 12V voltage regulators with carefully selected, high-grade components. And because the NCx500 OEM module can do more current, the rated power of 700W at 4Ω is now also available at 2Ω.

Another new technique implemented in this new module series is a differential idle current, which results in a lot less heat dissipation and reduces idle losses 20% compared to the NC500 OEM module.

The Danish Contenders

Of course, the competition is not standing still and at High End Munich *audioXpress* had the chance to see firsthand the latest two new amplifier modules from Purifi Audio, expanding on the company's extremely successful 1ET400A, 400W Class D single-channel, analog-input, amplifier module. The Danish company, founded by Bruno Putzeys, Lars Risbo, and Peter Lyngdorf, together with Claus Neesgaard, Søren Poulsen, and Kim Madsen, all coming from Texas Instruments, plus speaker experts Carsten Tinggaard, and Morten Halvorsen, is causing ripple effects across the audio industry with its bold approach to pushing the industry standards.

Increasingly, the Purifi team is adopting an integrated approach to perfect the audio amplifier circuit, pushing Class D to its limits of performance, while designing speaker drivers that are able to translate that power performance into sound. Purifi's Eigentakt amplifier technology nearly eliminates harmonic and intermodulation distortion and noise while its load-invariant frequency response can

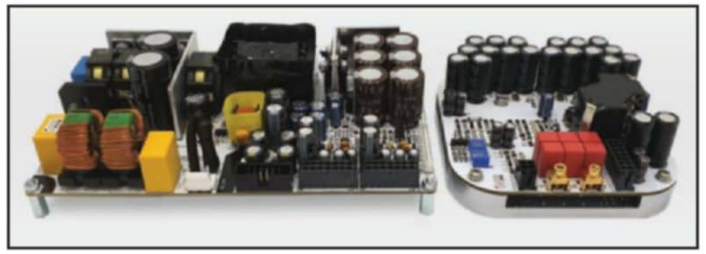


Arylic Up2Stream Amp V4 stereo amplifier board, a popular solution for DIY and integration, is available for $119.00 USD and already supports wireless audio multiroom streaming.

The Hypex UcD700HG amplifier module is a self-contained high-performance Class-D amplifier, available in the DIYclassD UcD700 mono kit, an easy way to get a high-end hi-fi system using the HxR technology for a very affordable price.

drive even the most demanding of loudspeakers to their highest performance. And since they achieved wide industry recognition and commercial success with the existing 1ET400A amplifier modules and matching front end and control system, in Munich, Purifi unveiled what can be considered a logical expansion of that strategy.

First with the new 1ET7040SA Eigentakt, which was already known by some manufacturers and builders and had its world premiere at the High End show in Munich. Basically, the new module delivers much more current (40A) and power—950W (2Ω), 500W (4Ω), 250 (8WΩ)—while maintaining the same very low levels of noise and harmonic distortion. Available specifications for this new module indicate THD+N below 0.00035% (measured at 200W, 4Ω, 1kHz), with 129dBA dynamic range. Efficiency is rated at 94% (measured at 500W, 4Ω, 1kHz) with idle losses also greatly reduced at the output stage. But more important, is also the fact that Purifi has pushed the design to its limits in order to determine its behavior under maximum stress. And with that knowledge, the team designed all sorts of protection measurements for maximum loads, including current limitation, DC protection, and thermal protection plus under- and over-voltage protection. This is a solution that should be in the market by the time this issue of *audioXpress* is distributed, but as with all companies these days, volume production might be conditioned by components availability.



The latest in Class D for the high-end audio enthusiasts is the Hypex Nilai500DIY module and matching power supply, which are promising to offer specifications that are "10 times better" than the company's popular NCORE technology.

The new DIYclassD Nilai500DIY preamplifier, stereo amplifier, and mono block amplifier kits featuring the latest Nilai Class-D board and PS500DIY power supply, as shown at the High End 2022 show in Munich.

And still without a release timeframe, what is completely new from Purifi is the new 1ET9040BA Eigentakt amplifier module, using a BTL topology that will be able to deliver 1500W at 2Ω. The module also comes with a connector and control system, unlike the previous modules. According to Purifi, this will be an amplifier focused on the OEM market and is intended to take the company forward to new application levels that require much more power, while maintaining outstanding performance levels.

And since we are discussing higher power levels, another major player in the OEM amplifier market, ICEpower, just recently strengthened its pro audio lineup of modules with two new members. The ICEpower 2000AS HV (High-Voltage) is a fully integrated audio power amplifier module, including a universal mains power supply with PFC as well as ErP and Energy Star-compliant standby. The new amplifier module will be available with one amplifier channel of 2000W (2000AS1 HV) and two amplifier channels of 2000W (2000AS2 HV), both channels capable of 2000W into 4Ω and 1400W into 8Ω. According to the Danish company, the new 2000AS HV amplifier module builds on the legacies of the existing 700AS and 1200AS modules, as a power solution targeted for live sound experiences.

The 2000AS HV amplifier power modules combine an amplifier based on ICEpower's Class-D ICEedge technology, with a universal mains switch mode power supply with PFC. This includes ErP and Energy Star-compliant standby functionality, regulated auxiliary power supplies, wake-on signal sense and a DC-bus output.

As the Danish company explains, to increase output power and voltage, the transformers, output devices and output filters had to be upgraded, while the design kept the noise and distortion performance in check. Significant effort was also made to optimize thermal dissipation and improve the protection system as required by the large voltage swings and high peak power of live-sound applications. Of course, the new modules will be also ideal for high-end home subwoofers and commercial audio installations with 70V/100V content voltage lines. Both channels are capable of 2000W into 4Ω and 1400W into 8Ω. The use of a regulated power supply topology ensures consistent power performance globally across both low and high mains.

Improving Class-D Amplifiers

Not all companies are focusing on the audio amplification circuits and topologies only. With Class D reaching new levels of performance, there are many optimizations that can be extremely important for different applications, considering power efficiency, or audio quality from cost-competitive designs.

A company following that approach is Axign, a Dutch startup that has already seen significant success working with large clients such as Harman, and others that will possibly be revealed later in 2022, with major market wins expected for 2023. In previous Market Updates, *audioXpress* detailed one of those achievements with the Harman Kardon Citation amplifier, and more recently Harman revealed that the JBL 4305P Studio Monitor from its Luxury Audio division also uses the company's AX5689 chip. And there's probably more Harman products using the technology that have not been disclosed yet.

The reason why the company has quickly attracted the interest of the audio industry has to do with its patented audio technologies and market approach, basically offering a full audio bandwidth feedback loop that enhances the performance of any Class-D audio amplifiers. This first-in-class technology retrieves the signal after the output filters and processes it in the digital domain with a carefully designed digital compensation filter in a single loop that suppresses nonlinearities and frequency dependent load behavior in the LC output filter. The patented technology includes a low-latency converters and very high speed digital processing.

The overall signal-processing path consists of a digital input interface, volume control, an interpolation stage, the digital loop-filter and a PWM controller. The PWM signal directs the power stage with a reconstruction filter delivering a clean analog signal to the speaker. A true digital amplifier solution with balanced feedback from the loudspeaker wires.

The combination of an integrated DAC functionality, high-resolution digital amplifier control loop, and post-filter feedback is getting appreciation from the industry also due to its reduced footprint of the final designs and improved sustainability. This process can be applied to many existing Class-D amplifiers for many different applications, including direct integration in audio ICs.

For this market update, *audioXpress* spoke with Richard Langezaal, the company's vice president of business development, to learn a bit more about the direction of the company.

Axign is winning traction from various consumer and professional audio brands, which are using it in more and more amplifier and active speaker platforms. "New technologies always take time, but even the audio companies in the automotive industry are knocking on our door," he confirmed.

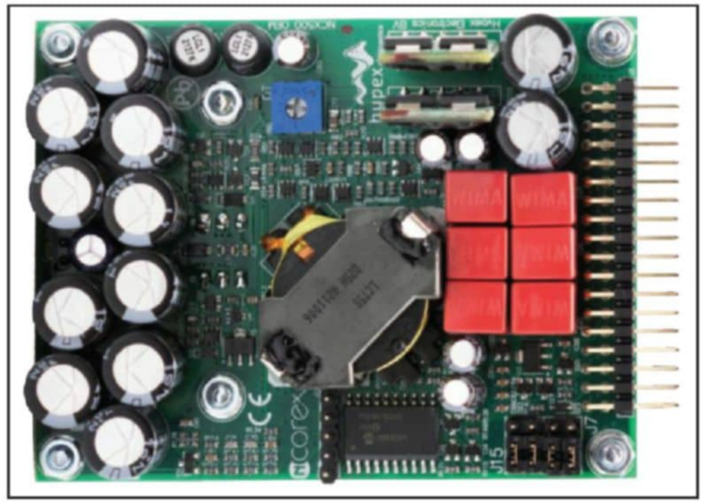
"Our post-filter feedback loop enables us to use more efficient ways to amplify audio. Ultra-high-efficiency removes heatsinks from high power audio amplifier solutions, or ultra-high-efficiency maximizes the battery playtime of portable audio devices. It does not matter if we use a new embedded power stage (lowest cost), a MOSFET power stage, or a GaN FET power stage. For all building blocks we have optimized solutions, depending on output power and system cost requirements."

Currently, Axign is offering two products. The AX5688, a two-channel BTL audio amplifier controller IC with digital inputs and CMOS level PWM outputs; and the AX5689, a four-channel with high-order digital control loops, and feedback after Class-D output filters. Those are being used in soundbars, streaming amplifiers, active speakers, smart speakers, subwoofers, and more.

And Axign is exploring new solutions, including working on a new generation of post filter feedback controllers in the voltage and current domains. As Langezaal explains, those will include functionalities "to process and adapt the complete signal path, stand-alone, in a dynamic supply environment, with more control over the loudspeaker characteristics." And for lower power applications, Axign is working on new solutions for headphones and earbuds that can improve on AEC and Adaptive ANC.

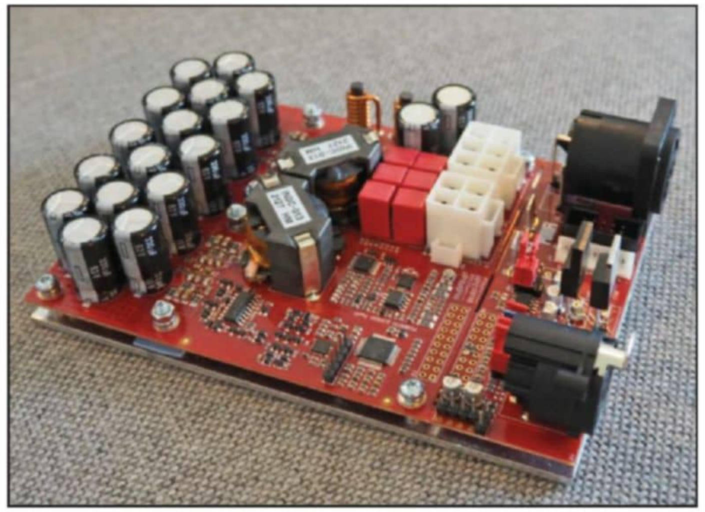
The Power of Gallium Nitride

As already mentioned, we cannot possible do a Market Update on audio amplification without discussing gallium nitride (GaN) technology. Currently all the major semiconductor companies, from Infineon to Microchip, are actively pursuing GaN solutions for audio applications, but their major focus is obviously power conversion and powertrains for electric-powered mobility.

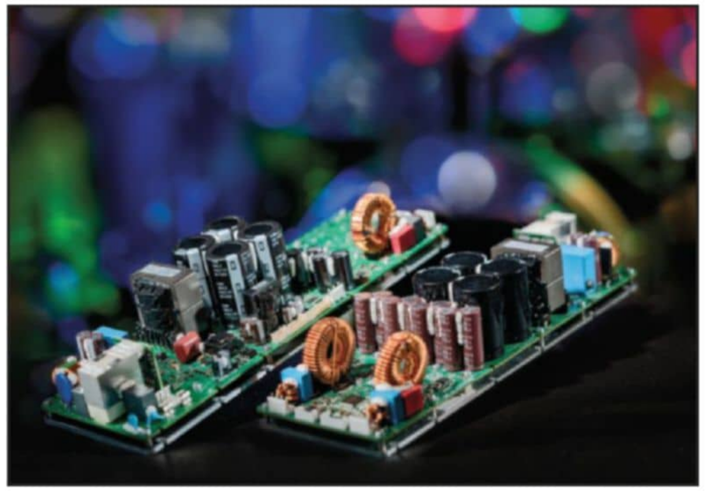


The big news for the OEM market from Hypex Electronics is the introduction of the new and improved NCOREx Class-D amplifier technology, of which the NCx500 OEM is the first new module to be launched. According to the Dutch company, the new technology will afford at least 2× better performance compared to original NCORE products.

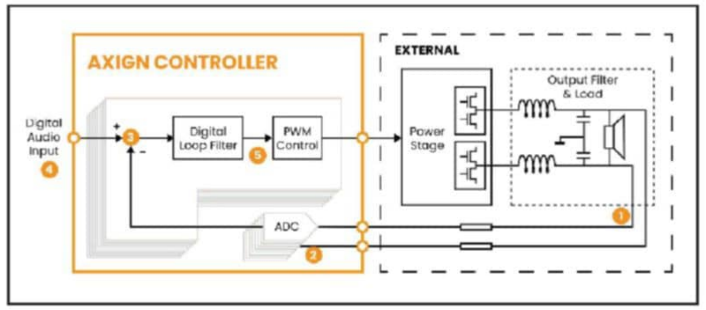
In 2022, NAD Electronics took value and audio performance to a whole new level with the launch of its new M23 Masters Series stereo amplifier featuring once again Purifi Eigentakt amplifier technology, manufactured by NAD under license from Purifi Audio. Continuing NAD's long-standing tradition in identifying and developing cutting-edge amplification technology, the NAD M23 is a stripped down version of its M33 Master Series BluOS Streaming DAC Amplifier, for nearly $1,500 USD less and no screen on the front panel.



The new Purifi Audio 1ET9040BA EigenTakt, the first bridge-tied load (BTL) amplifier module from the company.



The new ICEpower 2000AS1 HV and 2000AS2 HV fully integrated amplifier modules put out 2kW per amplifier channel, with 149V of available voltage swing.



The Axign approach consists of an analog signal measured at the loudspeaker terminals (1), digitized by a low latency ADC (2), into a 1-bit stream that is passed on to the digital signal processor (3). Here the signal is compared with the digital input (4) to generate a new signal that compensates for the errors, nonlinearities and noise in the path to the loudspeaker. In the PWM controller (5) the PWM signal is made to direct the power stage. The reconstruction filter is used as a low-pass filter to remove the high frequency content of the PWM signal.

The gallium nitride power semiconductor specialist currently leading in that space—and gaining an increasing margin with that success—is clearly GaN Systems, from Ontario, Canada.

With a new development platform and solutions, GaN Systems is reinforcing the value proposition for audio manufacturers, offering not only improved audio quality (which depends on the overall implementation) but also smaller audio systems, more power, and streamlined designs with the elimination of heatsinks. Adding to the company's recent success stories with products such as the Syng Alpha Cell speaker, Orchard Audio's groundbreaking Starkrimson Stereo Ultra amplifier, and the latest all-in-one Starkrimson Streamer, Ultra GaN Systems has made important announcements for audio developers.

Just recently, GaN Systems announced the launch of its 300W Audio Boost Converter SMPS, with a reference design, targeted for automotive, marine, motorcycle, and a broad range of 12VDC input applications, delivering improved audio quality and smaller, more streamlined audio systems.

The new 300W continuous, 500W peak power audio-grade SMPS reference design offers a complete solution with a wide-range DC supply voltage input of 9VDC to 16VDC, load dump rated, and operates seamlessly with GaN Systems ±32VDC Dual-Rail output amplifier. Along with wide range DC supply voltage input, onboard common-mode AC line filter, and differential primary drive transformer topology, the platform also provides SCP, OCP, thermal, voltage protection, and auto recovery.

This is complemented with an updated version of its Class-D audio amplifier evaluation kit, allowing evaluation for high-efficiency audio reference designs. Evolving from the original evaluation kit introduced in 2020, GaN Systems released the GeN2 Amplifier and SMPS Reference Design, shrinking the size and increasing its capabilities for Class-D audio designs.

This new solution is optimized for sound quality, thermal performance, size, and cost. The evaluation kit includes a two-channel, 200W per channel (8Ω) Class-D audio amplifier with 96% efficiency and companion 400W continuous, 550W peak power audio-grade switched-mode power supply (SMPS). The combined solution enables audio design engineers to create premium audio products more quickly, reducing time to market and at affordable prices.

The new reference design features a 20% overall size reduction in the power supply, has a higher level of protection into lower impedance loads to the amplifier and provides better thermal management resulting in 10°C temperature reduction. In addition to these enormous improvements, GeN2 also lowers bill of materials (BoM) cost. The solution offers high efficiency across a wide load range, and designers can quickly scale to a higher power with magnetics and GaN transistor selection.

Additionally, the solution highlights an easy plug-and-play design with features such as multi-audio signal inputs, bridge-tied load output, and open-loop/closed-loop toggling.

GaN Systems predicts an exponential growth of GaN in the Class-D audio market, and that this will be the year when the biggest audio names will launch GaN audio products.

As Rick Reigel, Vice President of Sales at GaN Systems states, "It's an exciting time in the audio industry where technological advances and consumer trends are driving rapid innovation and new user experiences. GaN is powering that innovation."

AudioControl, the audio amplifier manufacturer that was recently acquired by automotive aftermarket company AAMP Global, continues to expand its high-performance range of solutions for automotive, residential, and commercial applications, and recently embraced GaN technology from GaN Systems. AudioControl engineers created a circuit that they enthusiastically say establishes a new bar for power and sound quality.

The new AudioControl CM Series is available in four-channel, three-channel, and two-channel 70V Dual-Mode amplifiers. The 70V amplifiers are useful for driving many loudspeakers but have always suffered from performance limitations, often lacking clarity and reduced bass extension. AudioControl says that the resulting CM Series amplifiers are able to achieve "high-output power, unparalleled efficiency, reduced distortion and ultra-low signal-to-noise resulting in outstanding sonic rendering from a highly efficient circuit that fits elegantly inside a sleek form factor."

There was also another major announcement for the audio industry in 2022, which was the unveiling of a collaboration between Axign and GaN Systems in a new heatsink-less audio amplifier. The two companies produced a reference design that merges both companies' technologies, resulting in a new system design that reveals a "new world of extraordinary audio performance" in the company's words. By leveraging the technology developed by Axign, implementing a digital control loop with feedback behind the output filter, with the proven benefits of GaN technology, the solution signals the next leap in efficiency and sound quality performance. It is a design the two companies believe will be ideal for consumer and automotive audio applications, helping to solve technical challenges, especially heat generation in high-power audio amplifiers and efficiency, while staying within Class-A rated parameters at higher power.

While originally rated as a 500W, GaN solution (2× 250W at 4Ω) when it was announced for CES 2022, the design has been meanwhile upgraded to 1000W bridged or 2× 500W at 2Ω. As Langezaal confirmed to *audioXpress*, "the power losses in the GaN FETs are well below 1W in total, in idle mode and the whole board remains cool with worst case music output power. No heatsink is needed. And that is exactly what we want. Most power needs to go from the power supply into the loads."

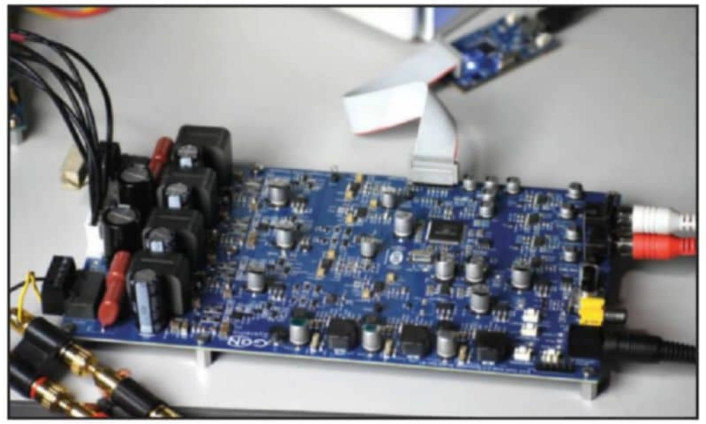
And other companies are exploring other approaches with GaN technology. As *audioXpress* author Stuart Yaniger found out during his experiments with GaN Systems evaluation audio board: When driven in open-loop mode, the signal could be made to sound almost as a tube amp. And that's precisely what AGD Productions, a small boutique audio company from Los Angeles, CA, is exploring in its range of high-end power amplifiers. As Alberto Guerra, the company's technical leader and founder defines, aiming to deliver "the best possible experience in the reproduction of music."

AGD launched the Vivace, and the Audion monoblock amplifiers, and recently introduced the Tempo stereo amplifier, all using the same Gallium Nitride power stage. The difference in the company's approach resides in its unique GaNTube design, which explores that "tube sound" relation of a GaN stage, but openly evokes that association to the user, by placing the power stage circuit enclosed in a glass tube (illuminated with LEDs to simulate the effect). Of course, there's much more to it in each model, but we will leave the details to the manufacturer's website.

Multichannel for Immersive

With highly efficient Class-D audio amplifier modules, the audio industry is now deeply committed to fully embracing a transition from conventional stereo directly into multichannel audio, in order to support immersive/spatial audio experiences.

And of course, immersive audio starts at the source, in the studios where content is produced and prepared for distribution or presentation in a live performance. Naturally, all studio monitor



GaN is being enthusiastically embraced in audio product development, enabling manufacturers and brands to launch better sounding, higher-performance, smaller, and more efficient audio systems. And GaN Systems is making it easier to reap the benefits of GaN with the introduction of the GeN2 Amplifier and SMPS Reference Design.

Following the enormous success of the original Starkrimson Mono amplifier, Orchard Audio introduced the Starkrimson Stereo Ultra amplifier, using the same proprietary dual-feedback modulator and next-generation GaN Systems FETs. Starkrimson amplifiers use Leo Ayzenshtat's proprietary DC-coupled, fully balanced dual feedback modulator, which allows the amplifier to be completely balanced from input to output, through the use of bridged GaN power stages. A great design and implementation of the technology by Leonid (Leo) Ayzenshtat, Orchard Audio's founder, that was recently reviewed by *audioXpress* (May 2022).

A new groundbreaking GaN-based heatsink-less audio amplifier is the result of a cooperation between Dutch company Axign and GaN Systems, resulting in a new system design that combines very high power with all the benefits of Axign's digital control loop with feedback behind the output filter, across the loudspeaker terminals.

manufacturers are dealing with those requirements, and particularly those that do not promote active designs have felt the need to leverage the latest available modules to respond.

An example is Finnish loudspeaker manufacturer Amphion that has launched its Amp400.8 amplifier in response to the rapidly growing immersive audio market and proliferation of Dolby Atmos audio productions in film, television, gaming, and now also music. The Amphion Amp400.8 matches the company's full range of passive studio monitors with eight identical Class-D channels. Combining the latest advanced Class-D technology with an in-house designed proprietary buffer stage, the Amp400.8 offers up to 410W of clean, dynamic power across up to eight channels.

Major requirements in the design was the need to provide coherence between all of its eight Class-D channels, giving it consistent power and performance for any multichannel setup. Naturally, audio companies focused on cinema and home-theater applications have naturally benefited from this transition but are all feeling the need to update their solutions to the latest technology.

AGD Productions Audion and Gran Vivace amplifier monoblocks, using a GaN power stage enclosed in a glass tube—the GaNTube power stage—that the company selected for its ability to efficiently switch at much higher slew rates than any silicon based power MOSFET, with almost perfect behavior and oscillation free switching.

Even NAD Electronics recognized the trend and the Lenbrook company introduced the M28 seven-channel amplifier, bringing the Purifi Eigentakt technology to the home theater, already adopted in the highly praised all-in-one NAD M33 Master Series BluOS Streaming DAC Amplifier.

Trinnov Audio, the French designer and manufacturer of reference audio processors for home theaters, high-end hi-fi, professional audio, and commercial cinemas, announced the release of the Amplitude16, a new high-channel count power amplifier. Designed for immersive home theaters, with custom integration in mind, the Amplitude16 offers a versatile power amplifier option, providing integrators with the unique ability to meet diverse channel counts and power requirements with exceptional rack efficiency.

Designed in partnership with ICEpower, according to Trinnov specifications, the Amplitude16 maximizes the performance of the latest Class-D modules, to enable a design that is effectively two eight-channel amplifiers on a single 3RU chassis, with dual power supplies, each requiring its own 15A AC mains power inlet. In its base model, it is a 16-channel power amplifier with a continuous power rating of 200W/channel at 8Ω and 400W/channel at 4Ω. Maximum continuous power with all 16 channels driven is 2800W at 115V and 3200W at 230V with less than 0.1% THD+N. Peak Power with all channels driven is 6080W with 1% THD+N. In bridged mode, the power for one channel increases to 800W/channel at 8Ω and over 1,000W/channel at 4Ω.

The input sensitivity of the Amplitude16 is optimally matched to the output level of Trinnov's Altitude audio-video processors, maximizing the system's SNR and dynamic range.

Trying to circumvent worldwide supply chain issues, Pro Audio Technology (PRO), the manufacturer of high-output loudspeakers and DSP loudspeaker controllers, also decided to create a dedicated line of modular multichannel amplifiers. Earlier in 2021, Paul Hales, president and product designer for Pro Audio Technology decided it was time to plan for the worldwide shortage of digital and other electronic components, which was already impacting the industry's ability to deliver complete systems. The company originally announced the MA-4242, MA-9999, and MA-9900 modular amplifiers, and the new range is ready and available also with the MA-4400 and MA-9942 models, all of which can be combined with PRO's loudspeaker controller/processors. All are built on proven Class-D technology from ICEPower.

The Professional Audio Amplifier Market

Quickly jumping to professional audio applications, there is obviously a lot of progress in terms of the use of technology that is provided, either via OEM suppliers or developed in house. But there's also a noticeable trend in the industry to reinforce the alignment between professional audio amplifier companies and speaker companies. This is due to two major factors. One is related with the integration level that exists in pro audio, where the speaker processing and crossover presets are directly stored in integrated amplifiers, which are distributed and networked. Second, because of supply chain issues, the speaker companies want to establish strong links with amplifier vendors in order to benefit from the latest innovations but more importantly to ensure a steady supply.

And the large groups in audio know this better than anyone and are acting decisively when required. A significate example of that was the recent acquisition of Linea Research Holdings, by Focusrite plc, the British music and audio conglomerate company that also owns Martin Audio among other brands.

Focusrite recently created Optimal Audio, a new installation-focused brand that naturally also needs amplifiers and processors for its speakers. Linea Research is a UK-based designer and manufacturer of high-power Class-D amplifiers and DSPs, with a strong knowledge of audio networking and software products, including remote monitoring. Headquartered in Letchworth Garden City, UK, Linea Research was formed in 2003 by a team of experienced professional audio specialists that previously led the product development efforts at Harman's BSS Audio, and were responsible for creating world renowned products, such as the Soundweb and Omnidrive processor range, or the Varicurve multiband parametric equalizer.

Linear Research current product portfolio includes a range of professional audio amplifiers, including the M Series of high-power, multichannel Class-D amplifiers with advanced DSP and network control. Focusrite's subsidiary Martin Audio is a significant customer of Linea, which is behind its successful iKON amplifier series.

However Focusrite also confirmed the intention to continue to develop the sales of Linea products through third-party distribution and to OEM loudspeaker customers, which include many famous professional loudspeaker companies in the UK, Germany, Italy, Spain, and Brazil, including until recently Danley Sound Labs in the US.

In a premonitory move, Danley Sound had signed an agreement in September 2021, to use Dynacord equipment, part of Bosch Communications Systems, teaming up to bring complementary product combinations to the global market. Danley loudspeakers will now be powered and optimized by the latest generations of Dynacord amplifiers, digital signal processing, and sound system software. And Danley loudspeaker settings will be available for all Dynacord amplifiers.

Dynacord is known for high-quality, reliable amplification and active sound systems, with hundreds of thousands of amplifiers made by Dynacord used around the world. In recent years, Dynacord stepped up its technology benefiting from developments in Class-D amplification, digital signal processing and networked audio.

Recently, Dynacord introduced a key development in networked audio, with the introduction of the MXE matrix mix engine series, providing comprehensive routing and mixing capabilities, while also serving as a communications hub for IP-based peripherals with extensive control and monitoring functions.

In a similar move, following the acquisition of Linea Research, Fulcrum Acoustic and Belgian amplifier manufacturer APEX have announced an exclusive distribution partnership that will see APEX CloudPower installation amplifiers introduced to the US market. Fulcrum recognized the powerful combination of pristine audio performance with the ability to control amplifiers remotely of CloudPower amplifiers.

CloudPower is a family of four-channel Class-D amplifiers, available in four models ranging from 350W to 3000W per channel, built on APEX proprietary GlidePath direct drive architecture. Multiple

GaN-powered AudioControl CM Series 70V Dual-Mode amplifiers are further examples of a best-in-class, innovative product design leveraging the benefits of GaN Systems technology.

amplifiers can be easily monitored and controlled remotely via the APEX Cloud using a web user interface. CloudPower is also the first amplifier in its class with an integrated audio source and streaming capability, allowing easy streaming of background music from devices running apps such as Spotify or AirPlay.

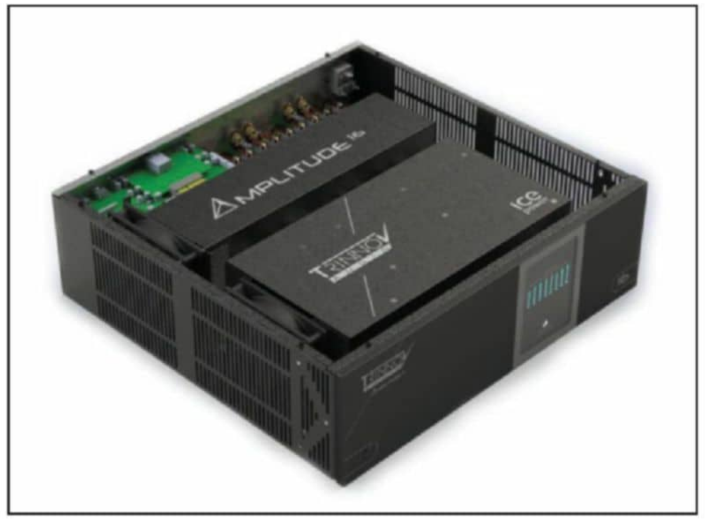
Installation Amplifiers

Also showing signs of radical market changes, the installation market has been inspiring audio amplifier companies to expand their dedicated offerings, sometimes even leading to branding reorganizations.

An example just recently confirmed comes from Pascal Audio, the Danish audio amplifier company that always assisted pro audio equipment manufacturers with dedicated amplifier electronics and OEM support. Its highly advanced and powerful compact OEM amplifier modules for the professional PA loudspeaker industry made it a leading name in amplification.

ISO 9001 certified, Pascal combines its R&D, product planning and engineering operations at its Copenhagen headquarters, while manufacturing is outsourced from China. It was just in 2017 that Pascal separated from Camco, the pro audio amplifier company,

Finnish loudspeaker manufacturer Amphion has launched its Amp400.8 amplifier designed specifically for the needs of multichannel studio monitoring.



Designed to Trinnov's specifications in partnership with ICEpower, the Amplitude16 maximizes the performance of their Class-D modules with custom dual power supply and heat dissipation design.

and became fully controlled by its Danish founding team, following the acquisition of Camco by L-Group, the holding company for L-Acoustics, in November 2016. And the main reason for the Pascal buyout was identified as the need to be a fully independent OEM provider of amplifier solutions to the pro audio industry "fully autonomous of potentially conflicting house branded product," as CEO and co-owner, Lars Fenger stated then.

Of course, L-Acoustics remains a major client for Pascal, using the company's patented technology and designs in all of its Class-D based amplifiers. All Pascal amplifier module designs feature proprietary core technologies, including its patented UMAC Class–D amplifier platform and UREC universal power supply design that enables universal AC mains operation.

Now, in 2022, Pascal seems to have had a change of mind and recently announced complete amplification systems available under its own separate brand, Blaze Audio. This remains for now a strategy mainly focused on the fast-evolving installation market, where amplification became a central point for fully networked distributed systems that combine controllers, processors and signal routing.

As Pascal explains, the new range of Blaze Audio amplifiers is based on the new Pascal IP series and smart amp technology, which combines full chassis with two or four channels with power share, full matrix DSP, TCP/IP control with zone and speaker management, Wi-Fi access and other features, available from Pascal with customization and branding options.

In March 2022, Linea Research joined the Focusrite Group of companies but committed to continuing existing partnerships with its long-standing OEM partners. Pictured is Linea's 44M Series of four channel amplifiers, optimized for live sound applications with integrated DSP.

At the recent ISE 2022, Pascal announced Blaze Audio as a subsidiary of Pascal A/S, entering the field with the PowerZone Connect series of fully matrixed, DSP-loaded network power amplifier products—basically a rebranding of the existing IP series products. And the declared goal is to convert Blaze Audio into "a major commercial and pro audio provider," selling directly to AV integrators.

The inspiration for this change of strategy was obviously motivated by the exponential growth of companies such as LEA Professional in the installation market. The manufacturer of cloud-connected amplifiers for professional audio applications was launched before the pandemic, and quickly conquered an important market share, given the vast knowledge of the market accumulated by its founding team, but most of all because of the fact that they introduced exactly the right product at the right moment. And the global pandemic confinements just accelerated that successful combination.

The South Bend, IN-based company didn't stop perfecting and updating its products in the last two years, and vastly enhanced the cloud interface experience based on feedback from actual users of the product. Firmware upgrades continuously improved the features required to support both the users and system integrators, making the use of a Web UI and Cloud control a huge competitive advantage for LEA Professional

And the company also enhanced its security features with the introduction of a new 802.1X security protocol—the Institute of Electrical and Electronics Engineers (IEEE) Standard for Port-Based Network Access Control that provides protected authentication for devices on the network and provides secure network access—on the company's full range of Connect Series amplifiers. 802.1X is the gold standard of network authentication security and defines authentication controls for any user or device trying to access a LAN or WLAN.

Also significant was the fact that LEA Connect multimode amplifiers have seen significant adoption also for larger residential systems and custom integration projects, which accelerated during the pandemic. This gave the company a significant market boost, converting what was originally designed as a commercial-grade audio amplifier range into a solution also suited for small to medium-scale installations.

The ability to leverage the built-in Wi-Fi access point or the FAST Ethernet to connect to any local area network, plus cloud connectivity, allows the level of remote control, monitoring, configuration, and personal device access that is required in smart home systems. Another major factor was the high channel count that is increasingly required for these whole-home audio installations, also featuring entertainment spaces supporting immersive audio. The LEA amplifiers are multichannel units that can be freely stacked and configured for such requirements.

But of course the main reference for the new Pascal strategy results from the formula used by Italian manufacturer Powersoft, which always delivered complete systems to the audio industry with great success, side by side with a relatively smaller OEM amplifier module business. And with the evolution of amplifiers toward distributed and networked systems, with control and processing features, that model became even more important. Many professional audio companies buy and resell Powersoft amplifiers instead of rebranded chassis, because they consider that combination to be a competitive advantage, given the recognition that Powersoft gained in the pro audio industry.

The Italian company was always regarded as a pioneer of Class D, taking the topology to completely new high power levels, but more recently benefited largely from the integration of other features and software control in its latest generation models. As an example, its T Series of touring and live sound amplifiers continues to expand and recently gained two new T902 and T904 1U models with even higher output possibilities, making it possible to replace Powersoft legendary K Series, introduced in 2003, and awarding Powersoft the recognition as one of the most reliable and widely used amplifiers on the market.

But reliability, power density, and efficiency, are no longer sufficient, the Powersoft T Series, introduced in 2019, redefined amplifiers for live sound applications, adding daisy-chained Ethernet connectivity, Dante audio networking, power sharing, improved DSP with powerful Advanced EQ groups, and FIR filtering for great sounding presets, among other arguments.

The latest T902 and T904 amplifiers take the T Series even further, allowing it to cover the requirements for any system deployment with more power, improved onboard DSP Raised Cosine, IRR and FIR filters, delay, live impedance monitoring, as well as Powersoft's Active DampingControl for cable resistance compensation.

On the installation side, where Powersoft enjoys success with its Mezzo compact range of smart amplifiers, plus its powerful family of Ottocanali, Quattrocanali, and Duecanali 19" rack amplifiers, available with and without DSP or Dante, the Italian company has been able to quickly adjust to evolving market requirements due to its strong ArmoníaPlus software foundation that is being constantly refined and updated.

One of those areas of improvement was obviously on cloud integration. Powersoft recently announced that version 2.2 of its ArmoníaPlus software update now enables integrators to share projects and work collaboratively in the cloud. Primarily designed for systems engineers, and available in both Install and Live Sound front-ends, the update allows users to save projects in a Microsoft-based cloud, in addition to on a local drive. Also introduced with version 2.2 of ArmoníaPlus is the ability to monitor the status of Powersoft amplifiers via Simple Network Management Protocol (SNMP), ideal for fixed installations with a central monitoring platform. SNMP is available for all Mezzo, X Series, T Series, Quattrocanali DSP+D Series, Duecanali DSP+D Series, and Ottocanali DSP+D Series Powersoft amplifiers.

Via software updates, Powersoft has also been expanding its zone management tools for its Mezzo amplifiers, which also now benefit from the company's Dynamic Music Distribution (DMD) platform. Based on the dynamic routing capability of audio signals in Powersoft amplifiers, integrators are able to use Powersoft amplifier platforms to replace most of the outboard required to manage, shape, and share audio sources throughout the various zones in a variety of dynamic music distribution applications by merging a routing matrix, networking capabilities, and digital signal processing between multiple amplifiers in different locations. Powersoft's ArmoníaPlus system design delivers an object-based approach that enables integrators to easily configure the routing of sources and zones. It also enables the creation of different system configurations to be recalled, as well as to configure UI system controls with a pre-defined layout that doesn't require any programming, or to create custom ones.

Pascal IP Series of installation amps has created a foundation for the Danish company to create Blaze Audio as a separate brand focused on the installation market.

Blaze Audio PowerZone Connect is a series of ultra-compact, full matrix DSP amplifiers with built-in Wi-Fi that are configured and controlled from a smart Control app, enabling multi-zone sound systems for a wide range of commercial install solutions.

Powersoft latest T902 and T904 amplifiers expand the T Series family and feature onboard DSP, built-in Dante, and live impedance monitoring, as well as Powersoft's Active DampingControl for cable resistance compensation, with all models being able to deliver the same high output voltage.

d&b audiotechnik's latest D40 Class-D amplifier is an optimized version of the original 40D installation amplifier with eight-inputs to four-channels featuring advanced voltage management, DSP, and network control and monitoring via AES70/OCA.

Powerhouse Multichannel

An overview of audio amplification in 2022 wouldn't be complete without mentioning the latest products from two market leading companies in the professional audio space.

In 2021, d&b audiotechnik introduced its new D40 mobile amplifier, an optimized version of the original 40D installation amplifier. This four-channel Class-D amplifier with reduced size and weight, features advanced voltage management, and increased system performance, significantly reducing power consumption for improved environmental friendliness. This makes the D40 more economical for touring, rental and portable applications.

The D40 incorporates up to eight input channels with four analog inputs and four AES3 channels with corresponding link outputs. Each input channel can be routed to any of the output channels. The new model follows similar design guidelines from the existing d&b D20 and D80 amplifiers, sitting in between those models in output power but with added flexibility and efficiency. With a rich user interface and color touchscreen, the D40 uses DSP to incorporate loudspeaker configurations and user-definable setups, equalization and delay functions.

And of course, all features and monitoring can be fully accessed remotely. Access to the d&b Remote network is via Ethernet using the Open Control Architecture protocol (AES70/OCA). The D40 is controlled using the integrated web interface, which enables access via a browser, or using the d&b R1 Remote control software.

As part of the d&b family of networkable products, the D40 benefits from the full range of d&b technology software solutions including ArrayCalc, ArrayProcessing, NoizCalc and the R1 Remote control software. These elements all come together in the highly efficient workflow, maximizing the performance, usability, and value of these systems from planning and simulation all the way to performance.

Already at the latest ISE 2022 show, d&b audiotechnik introduced its new 5D amplifier with integrated Dante audio networking. An installation-focused compact design, the 5D is a four-channel DSP amplifier featuring flexible output power sharing—ideal to power installations with large numbers of smaller speakers, such as required to power its own immersive audio technology, d&b Soundscape.

The 5D is the first d&b amplifier with integrated Dante networking. This allows for flexible system configuration changes and convenient deployment of decentralized system designs.

With a more ambitious announcement, designed to create a completely new amplification foundation, L-Acoustics announced its new LA7.16i multichannel amplifier, described as an amplified controller because it basically conveys the intelligence required to predict power and load requirements from all channels, surmounting the limitations of current high power amplifier configurations used in most projects. Unique in the market, LA7.16i offers an unprecedented 16x16 architecture—each of the 16 output channels delivers up to 1300W at 8Ω or 1100W at 4Ω—in a 2U chassis with seamless networking redundancy.

Based on extensive research, the LA7.16i surmounts what L-Acoustics describes as "the limitations of current high power amplifier configurations" by applying L-SMART technology, an acronym for L-Acoustics System Modeling Adaptive Resource Technology, a controlling system that monitors audio signals and manages all outputs in order to stay within the available power from the supply, whenever the system demands more power than can be provided.

With this suite of advanced power management technologies, designers can mix and match any L-Acoustics loudspeaker onto one single amplifier. L-SMART uses proprietary predictive modeling algorithms to manage the power supply unit (PSU) and the individual amplification channels. Hardware sensors within the amplifier supply real-time information, which is analyzed by the DSP to dynamically match loudspeaker system needs. The PSU can then provide extremely high short-term peak power or lower power over longer hold times. This energy is delivered intelligently to the advanced Class-D output stages, assuring optimum system performance at all times. LA7.16i can drive any L-Acoustics speaker up to K2, enabling high-resolution amplification and processing of large-format speakers with less system budget.

The new L-Acoustics LA7.16i amplification system improves flexibility and efficiency in multichannel scenarios with its unprecedented 16×16 architecture in a 2U chassis, with each of the 16 output channels delivering up to 1300W at 8Ω or 1100W at 4Ω, with seamless network redundancy.

The innovative combination of high-channel density and power capability makes LA7.16i a perfect partner for medium to large permanent installations or immersive audio installations requiring a significant increase in discrete processing and amplification.

Resources

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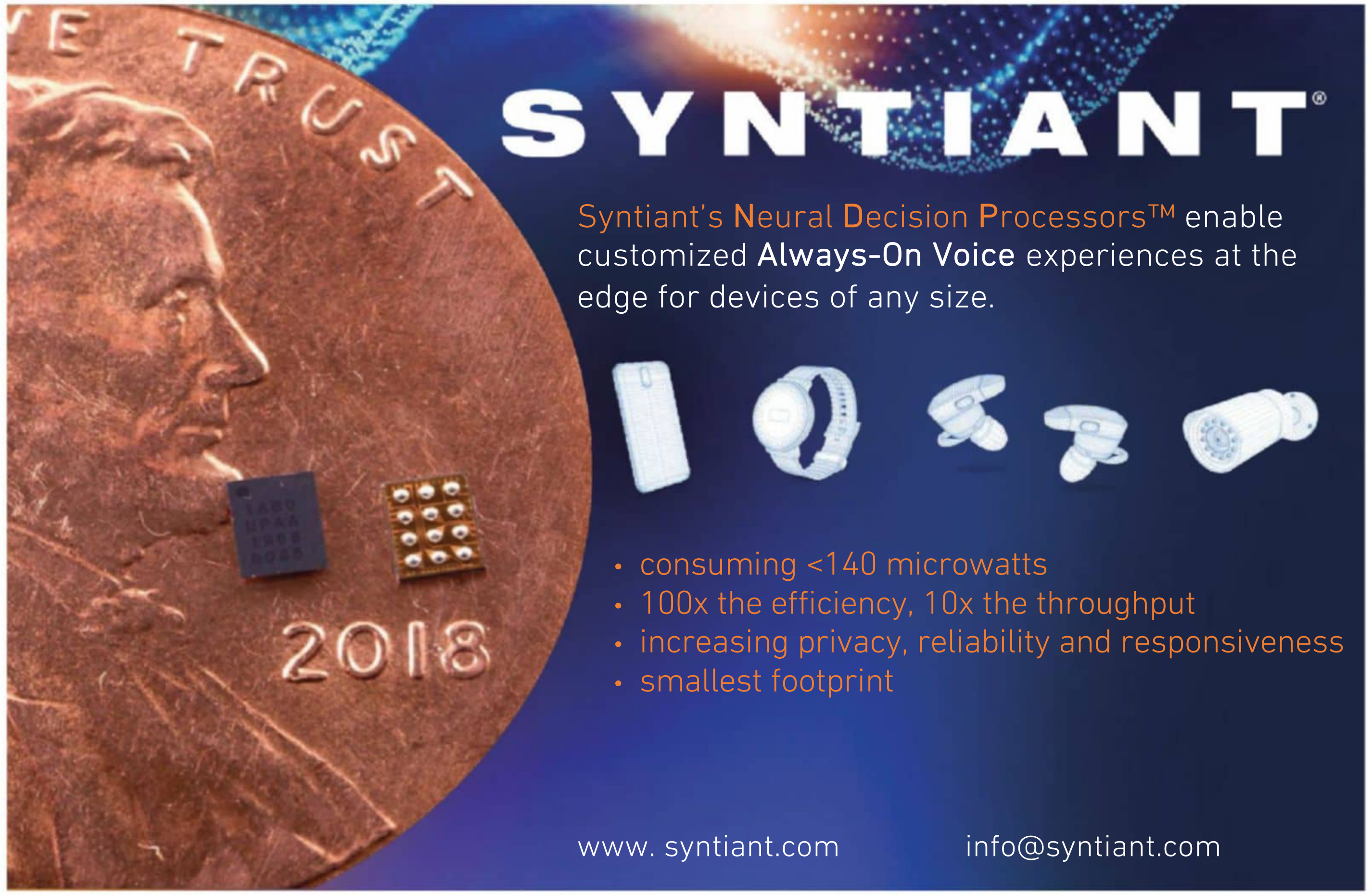

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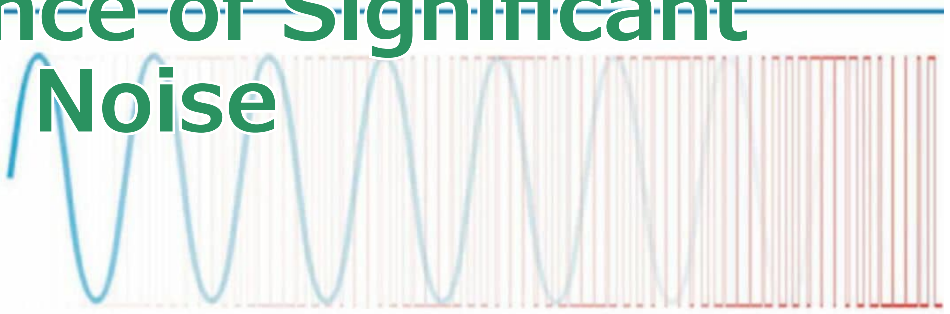


By
Bruce E. Hofer
(Audio Precision)

This article, written by the co-founder and Chief Analog Designer at Audio Precision and published in 2005, remains 95% on target for 2022. As the latest generation of Class-D audio amplifiers challenges our standard analyzers' ability to measure, Bruce Hofer already debated the fundamentals with the early instrumentation of the time. As a side note, Stuart Yaniger conducted some reference measurements.

There once was a time (before the 1980s) when the word "audio" referred to an analog signal having virtually all of its energy content below 20kHz. Any energy above 20kHz consisted of residual wide-band noise, signal harmonics, and perhaps the unintentional pickup of local radio and TV broadcast stations.

Audio analyzers typically provided one or more bandwidth limiting filters to reduce the effects of out-of-band energy on measurement results. These filters were usually three-pole in design with an 18dB/octave roll-off. The most common selections were 80kHz, 30kHz, and/or 22kHz; and few problems were ever encountered.

But times and technology have changed! The word "audio" has now come to include signals that can have gross amounts of energy just above the audio band. For example, the noise shaping commonly employed in most sigma-delta digital-to-analog converters (DACs) and the pulse width modulators (PWMs) of many Class-D amplifiers gives an exceptionally low noise floor within the 20kHz audio band; the trade-off, however, is an out-of-band noise floor that rises very rapidly just above 20kHz. Out-of-band energy may also include high-frequency artifacts and fast slewing transients related to the over-sampling converter or the amplifier modulator.

The traditional three-pole bandwidth limiting filter is virtually useless for rejecting these forms of out-of-band energy. Indeed, the AES17 standard for testing DACs specifies a very sharp low-pass characteristic that can be realized only with an elliptic filter of at least seventh order. This type of filter is much more costly and difficult to implement, and it can contribute higher levels of its own distortion and noise to the measurement. However, regardless of its design, the tacit assumption is made that all stages in the analyzer prior to its bandwidth limiting filter will continue to respond linearly to both the audio signal and the undesirable out-of-band components. This can be a very wrong assumption!

Where Is the Bandwidth Limiting Filter?

The architecture of a typical high-quality audio analyzer consists of the following blocks:

1) A selectable high impedance input attenuator to enable the measurement of large amplitude signals. Typical attenuator step sizes are usually 10dB or 12dB for a total range of 0dB (no attenuation) up to perhaps 36dB to 40dB in total attenuation.
2) A high impedance differential input stage having selectable gain. Typical gain increments are usually 5dB, 6dB, 10dB, or 12dB for a total gain range of 0dB to 36dB (or even higher in some designs). The purpose of this stage is to amplify small signals to the point where the noise contributions of following stages become insignificant when compared to the residual noise floor of the input stage.
3) One or more sets of auto-ranging comparators and means for automatically controlling the input attenuator and input stage gain for optimum performance.
4) An analog signal processing stage, such as a tunable notch filter, followed by additional gain stages, bandwidth limiting filters, and measurement detectors.

or

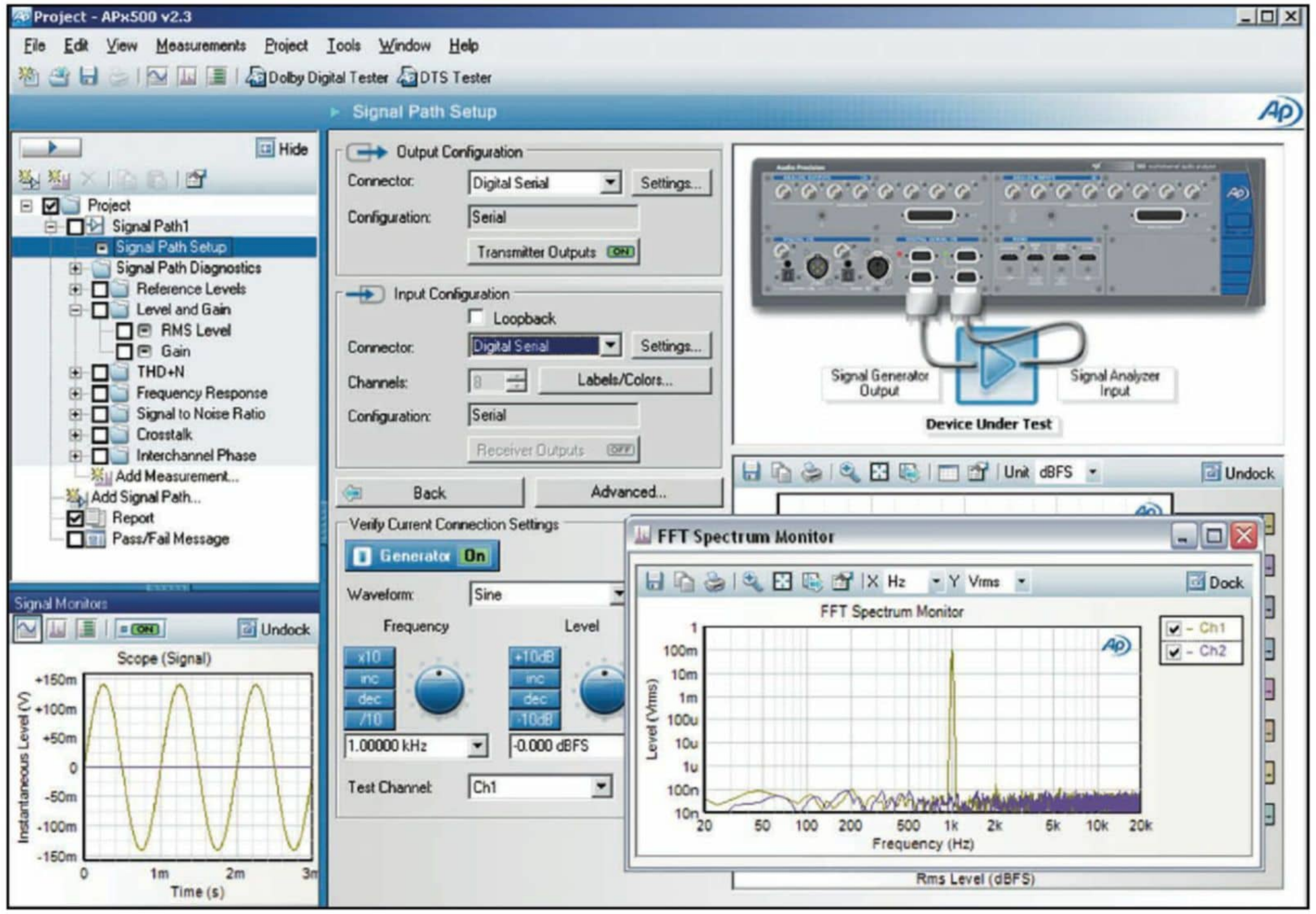
A high-performance analog-to-digital converter (ADC) followed by DSP processing to extract the desired measurements. A/D converter inherently limits the alias-free measurement bandwidth to <50 % of its sampling rate. Additional bandwidth limiting is implemented via DSP processing.

Regardless of how the last block is implemented, it is important to note that measurement bandwidth limiting occurs at or very near the end of the overall signal path. This has the advantage of limiting residual noise contributions from the analyzer itself. However all stages before this point in the analyzer must still process the full bandwidth input signal, including any out-of-band energy it may contain. As long as the out-of-band energy does not (1) exceed the amplitude of the in-band signal; or (2) contain high-frequency components that provoke slew rate limiting, the measurement system will behave predictably and linearly.

Auto-Ranging and Measurement Dynamic Range Problems

If the out-of-band energy exceeds the amplitude of the in-band audio signal, the analyzer auto-ranging circuits will no longer pick the same input attenuator and measurement path gain states as if there were no out-of-band energy. The in-band audio signal will be lower than its optimum level within each stage of the measurement path causing degraded residual noise performance and increased error. Depending upon the relative amplitude of the out-of-band noise compared to the in-band signal, the effects can be profound.

For example, suppose an audio analyzer is attempting to measure a 10mV, 1kHz signal in the presence of 1V of out-of-band noise. The input stage auto-ranging circuits will seek to pick the range that optimizes the measurement of the 1V out-of-band



Audio Precision APx software analyzer environment with basic signal path configuration shown as it looked like in 2009. That was the year Audio Precision introduced a new ultra-high bandwidth analyzer option for the APx525 family of audio analyzers, extending the FFT (Fast Fourier Transform) capability all the way to 1MHz, with 24-bit resolution and 2.38Hz bin width, making it ideal for looking at out-of-band noise in Class-D amplifiers.

Range	Gain	Maximum Input SR
160V	-36dB	640V/µsec
80V	-30dB	320V/µsec
40V	-24dB	160 V/µsec
20V	-18dB	80V/µsec
10V	-12dB	40V/µsec
5V	-6dB	20V/µsec
2.5V	0dB	10V/µsec
1.25V	+6dB	5 V/µsec
600mV	+12dB	2.5V/µsec
300mV	+18dB	1.25V/µsec
160mV	+24dB	0.62 V/µsec
80mV	+30dB	0.31 V/µsec
40mV	+36dB	0.16 V/µsec

Table 1: Using the Audio Precision System Two as the example, this illustrative table lists the overall gain and maximum allowable input signal slew rate versus input range setting due to this factor.

noise, not the 10mV signal otherwise clipping and gross nonlinearity would occur. The result is the in-band signal is approximately 40dB lower than it would have been if no out-of-band noise was present.

At some point in the signal path the bandwidth limiting filter removes the out-of-band noise leaving just the in-band signal to be passed on to the detector or ADC for measurement. In analyzers with analog detectors, a 40dB lower than normal signal will seriously degrade measurement accuracy.

Most analog detectors have only a 40dB to 50dB usable dynamic range due to quantization and residual DC offsets in their circuits. This is roughly equivalent to reading an old-style galvanic meter where the signal is only 1% or 1/100 of the full scale of the meter. Although the bandwidth limiting filter has successfully removed the undesired out-of-band energy before being measured, the amplitude of the desired in-band component is now so small in comparison to the full scale of the detector that the resultant measurement is subject to large error. In the extreme, the detector may not even register a reading above 0 (or −999dB)!

The situation is much better in A/D-based analyzers because the DSP implemented "converter" or "detector" has a much broader dynamic range, typically well over 100dB. However, the converter noise floor, distortion, and spurious contributions will be 40dB higher compared to the signal amplitude than had there been no out-of-band energy in the first place. Numeric round-off and truncation errors within the DSP algorithms can also lead to increased error when the signal being measured is below its optimum level.

Measurement Path Nonlinearity and Input Slew Rate Problems

When the out-of-band signal contains high-frequency components beyond the specified bandwidth of the analyzer, the ability of the analyzer input stages to linearly respond to the total signal must be considered. The analog signal paths of all high-quality audio and FFT analyzers contain ultra-low distortion operational amplifiers or "op-amps," such as the AD797, the OPA627, or even the venerable 5534 to provide buffering, gain, and active filtering.

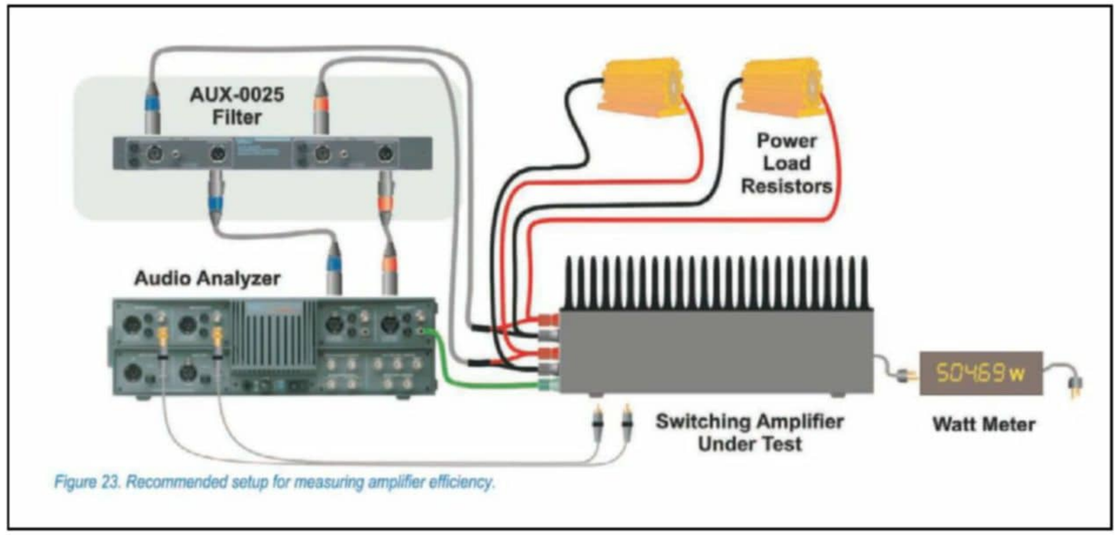
These devices have a typical maximum slew rate limit of 20V/µsec to 50V/µsec, depending upon their compensation. If the undesirable out-of-band energy components cause an op-amp to hit its slew rate limit, the in-band audio signal will no longer be processed in a linear manner. All subsequent measurements will then be subject to potentially gross errors.

For a given op-amp peak slew rate (SR), the maximum allowable output amplitude at a given frequency (or the maximum frequency at a given amplitude) can be calculated using the formula:

$\text{Peak SR} = 2\pi \times f \times (\text{Vrms} \times \sqrt{2}) = 8.89 \times f \times \text{Vrms}$ where SR is in units of V/µsec and "f" is in megahertz (MHz).

For example, the Audio Precision System Two analyzer uses AD797 op-amps in its input stage. The "full-scale" operating voltage at this point is 2.5Vrms. Given the slew rate of the AD797 is about 20V/µsec, the maximum full-scale signal frequency cannot exceed about 900kHz without hard slew limiting. Unfortunately the distortion performance of an op-amp degrades long before its slew rate reaches its limiting value. One design "rule-of-thumb" is to avoid signal conditions that push an op-amp beyond 50% to 60% of its maximum slew rate. Thus, the maximum rated full-scale frequency should not exceed ≈500kHz.

An equivalent way to think about this problem is in terms of an input signal slew rate limitation. Since the slew rate of an op-amp always refers to its output signal, the maximum allowable input signal slew rate will scale in inverse proportion to the gain (or attenuation) of the input stage. Thus the maximum input slew rate will necessarily decrease with increasing sensitivity (decreasing



A (2005) typical setup for measuring amplifier efficiency. If the amplifier is designed to operate from DC sources, input power can be measured by taking the product of the supply voltage and current measurements. Current can be measured directly or indirectly as a voltage drop across a known small resistor. If the amplifier operates from AC mains, measuring the input power poses a more significant challenge. There is simply no good substitute for a true power meter. The numerical product of an AC mains voltage and current measurement does not give power; it gives "VA."

The Audio Precision AUX-0025 / 0040 / 0100 switching amplifier measurement filters are designed to be inserted between the device under test (DUT) and an analyzer input, to reduce out-of-band switching signal components before measurement. The filters use a passive design for optimal performance, with custom inductors designed for high power handling and minimal low-frequency distortion. The AUX-0025 is a two-channel passive filter, with a 20Hz to 20kHz passband. It is designed to be used with two-channel analyzers, including the APx555, the APx52x Series, and the 2700 Series test systems.

voltage range). Using the Audio Precision System Two again as the example, **Table 1** lists the overall gain and maximum allowable input signal slew rate versus input range setting due to this factor.

This table is illustrative only, and it shows the limitation caused only by the input stage op-amps. The actual situation is somewhat more complex because there are other design factors that can more severely limit the maximum input slew rate, especially in the higher input voltage ranges. Similar tables can be constructed for other models and brands of audio analyzers based upon their choices for input voltage ranges, op-amps, and maximum rated signal bandwidths.

The key point is that all audio analyzers exhibit a decreasing ability to tolerate fast-slewing out-of-band components as the input voltage range is decreased. No amount of bandwidth filtering or processing in a later stage can repair the damage to the in-band signal if slew rate limiting has occurred in the input stage.

Input and External Filters

Audio analyzers often contain an internal radio frequency interference (RFI) filter in series with their input stages. A properly designed RFI filter can provide good rejection above about 5MHz and prevent input stage demodulation of FM and TV station pickup. RFI filters effectively increase the maximum allowable input signal slew rate at these higher frequencies. Unfortunately that still leaves the input stage susceptible to slew rate problems from out-of-band signals from just above the analyzer's maximum specified bandwidth up through about 5MHz.

Input or RFI filters can be designed to be effective down to much lower frequencies, but only with performance trade-offs that are usually unacceptable for general purpose audio analyzers. Such trade-offs can include a significantly higher input capacitance (or lower input impedance), higher input noise floor, degraded common mode rejection, and/or degraded flatness within the intended measurement bandwidth.

In certain applications where one or more of the above trade-offs is acceptable, a passive external filter can provide the needed rejection of out-of-band energy. External filters can be highly effective because they prevent the offending high frequency energy from entering the analyzer input stage in the first place. The Audio Precision AUX-0025 accessory for testing Class-D amplifiers is a good example. It is a dual-passive low-pass filter with a 20kHz usable bandwidth, very steep roll-off above 100kHz and a stop-band attenuation >50dB at 250kHz and higher. However, it also has an input capacitance of 10nF (10,000pF).

Testing DACs with Significant Out-of-Band Energy

The nature of D/A out-of-band energy varies considerably from manufacturer to manufacturer, and with different designs. General statements regarding the amplitude and the frequency distribution are difficult to make because of the wide range of noise profiles encountered. Some out-of-band signals are basically impulsive in nature, while others show the appearance of sine-bursts of one or more cycles at frequencies of several megahertz or higher. Although these signals can be filtered on chip, competitive pressures often force the omission of such filters.

The best tool to view these out-of-band signals is the digital storage oscilloscope. Looking at this signal using a spectrum analyzer can give very misleading indications. Significant impulsive or sine-burst forms of out-of-band signals can appear to be random noise if the time domain position of these artifacts varies in a random or pseudo-random fashion.

Because D/A out-of-band noise tends to be constant and relatively independent of the in-band signal, an interesting situation occurs when it is measured with an auto-ranging audio analyzer. As the amplitude of the in-band audio signal is decreased, the analyzer will switch to progressively more sensitive ranges thus reducing its input slew rate tolerance. Measurements of noise or THD+N will show a sudden jump or increase if the input stage gain switches to the critical point where the analyzer input slew rate capability drops below the slew rate of the out-of-band components. Once this happens, all subsequent analyzer measurements are subject to potentially serious error. One needs to be especially careful when making the Dynamic Range THD+N test at -60dBFS. Because the signal is so small, the analyzer will usually attempt to pick its most sensitive input range (with the lowest slew rate tolerance) if left in its auto-ranging mode of operation.

For example, a certain D/A has an out-of-band noise component that resembles a 4MHz sine-burst with an amplitude of about 60mVpp. The slew rate of this signal is calculated to be 0.75V/µsec. Using the table of input slew rate capabilities, it shows that linear analyzer operation can be expected down through the 300mV range of an Audio Precision System Two before input stage slew rate limiting occurs. Input range settings below 300mV should be avoiding when making measurements of this D/A.

As Daniel Knighten (AP, VP of Product Development) wrote in 2019, there are at least a couple of things that make audio signals and their measurement unique. "Practically speaking, a modern audio analyzer is asked to measure the DC offset of power amplifiers while observing the noise shaping and spurious out-band products emanating from Class-D chips and delta-sigma converters. Our own analyzers can resolve from DC to over a 1MHz with 1Hz resolution. Second, a modern audio analyzer needs to observe the output of everything from state-of-the-art D/A converters with noise measured in single digit microvolts (µV) to power amplifiers with 200 V outputs. Further, while measuring a 200 Vrms sine wave, the system must still be able to resolve the amplitude of harmonic products that may be 60dB to 100dB lower in amplitude than the fundamental. The APx555 has a self-noise of less than 1µV and a maximum input level of 300 Vrms, a range of 170dB. The requirement to measure signals with exceptional precision and accuracy over an incredibly wide frequency and amplitude range."

Resources

"AES Standard Method for Digital Audio Engineering — Measurement of Digital Audio Equipment," (AES17-1991), Audio Engineering Society, 1991.

B. Hofer, "Measuring Switch-mode Power Amplifiers," Audio Precision White Paper, October 2003.

B. Metzler, *Audio Measurement Handbook*, Audio Precision, 1993.

Testing Class-D Amplifiers

Class-D amplifiers operate by rapidly switching their outputs between two or more supply potentials using pulse width modulation techniques to control

About the Author

Bruce Hofer is one of the founders of Audio Precision and its principal analog design engineer. He remains technically active in spite of having become the company's majority shareholder and Chairman. Bruce's particular areas of design expertise include ultra-low distortion signal generation and processing, along with wideband linear signal amplification. Insiders have been known to call him the "precision" in Audio Precision because virtually every analog circuit in the company's product line bears his mark or influence.

Bruce's career with state-of-the-art instrumentation began in 1969. Prior to receiving his BSEE degree from Oregon State University in 1970, he was hired as an engineering assistant in the Lab Instruments Division of Tektronix under that company's summer student program. A year later he became a full-fledged engineer, and went on to design high-speed sweep generators and horizontal deflection amplifiers for Tek's famous 7000-series oscilloscopes.

In 1978 Bruce made a key career decision to respond to a growing personal interest in audio. He became the engineering manager and lead engineer of the Tek TM500 team that developed the world's first fully automatic distortion analyzer, the AA501 introduced in 1980. It was here that he also met or hired the other key team members that would ultimately leave Tektronix in 1984 to launch a new company called Audio Precision.

the power of the audio band signal. The raw output waveform consists of a series of variable width pulses whose amplitude is determined by the power supply rails. Typical switching frequencies are in the range of 250kHz to 750kHz. In the frequency domain, the raw amplifier output contains the desired in-band audio signal plus high frequency artifacts related to the switching frequency and its harmonics. The modulators of many Class-D amplifiers also employ noise-shaping techniques to favor low noise within the audio band. The resultant noise floor often resembles that of a sigma-delta D/A converter showing a rapidly rising characteristic just above 20kHz.

The amplifier's raw output waveform can have an incredibly high slew rate, typically 1000V/μsec or even higher. It is relatively independent of the actual audio signal itself. As can be seen from the table of input signal slew rate limits, there is virtually NO analyzer range setting that will process this signal without serious nonlinearity. Indeed, the slew rate is so high it will attempt to cause potentially damaging 200mA current spikes to flow into the 200pF input capacitance of the analyzer. Directly connecting an audio analyzer to the raw output signal of a Class-D amplifier is strongly discouraged!

To make valid measurements of Class-D amplifiers one must insert a suitable low-pass filter between its output and the analyzer input to attenuate the switching artifacts and their peak slew rate. Some Class-D amplifiers contain internal LC filters to minimize radio frequency interference. Although they dramatically reduce the amplitude of the switching artifacts and the corresponding peak slew rates, they may not be adequate to prevent input slew rate problems in all ranges of an audio analyzer. The Audio Precision AUX-0025 accessory referred to earlier was specially designed for this application.

Because most Class-D amplifier modulators employ noise shaping, the analyzer's AES17 low-pass filter should also be enabled to reject the effects of rapidly rising noise above 20kHz. Otherwise, making measurements on Class-D amplifiers is very similar to measuring a D/A.

Recommendations and Summary

Audio analyzers are optimized to make high-quality measurements up through their maximum specified bandwidths. The presence of significant energy above these bandwidths can cause serious problems with linearity within the input stages of the analyzer and introduce gross measurement errors. The test engineer must carefully consider the nature and amplitude/frequency profiles of out-of-band energy sources when setting up audio measurements. Unfortunately there is no automatic way to sense if out-of-band energy is compromising a measurement by provoking slew rate non-linearity within the audio analyzer.

Whenever possible, specify a fixed or minimum input range for analyzer operation when testing devices that have significant out-of-band energy. This prevents the analyzer auto-ranging feature from picking too sensitive a range where the input slew rate capability can drop below the actual out-of-band signal slew rate. If the slew rate of the out-of-band content is simply too high for any reasonable input range setting, an external low-pass filter (e.g., the AUX-0025) must be inserted between the device output and the analyzer input.

Good grounding and signal interconnection practices should also be observed to minimize common mode potentials (which can also have a very high slew rate) between the device under test and the analyzer inputs.

Some Notes—How Far Have Things Evolved?

By **Stuart Yaniger**

One downside to modern audio test instrumentation is the remarkable efficiency at collecting huge amounts of invalid data when attention is not paid to the details of test setup. Bruce Hofer's 2005 paper on testing Class-D amplifiers and DACs is still a valid cautionary tale of how ranging errors and slew limiting at an analyzer input from ultrasonic noise and artifacts is one of those details too often overlooked.

It's a tribute to Hofer's engineering insight that even after 17 years (an eternity in technology), his paper is still pertinent and significant for anyone involved in audio metrology. But technology does move at light speed, and both Class-D amplification and audio analyzer electronics are no exception. Audio Precision's APx series analyzers were introduced a decade later, with the B series updates following five years after that. The methods of voltage range selection in both hardware and software have evolved, and likewise, so have the input electronics (circuitry and devices) and their resistance to slewing distortion. Hofer's paper offers a table of maximum slew rates for each input voltage range on the old AP System Two. There does not seem to be equivalent data available for the APx series analyzers, but their higher bandwidth suggests that the maximum slew rates are likewise higher.

On the DUT side of things, Hypex NCORE, Purifi, GaN-based designs from Orchard Audio, and similar high-performance Class-D technologies have taken over the quality amplification market, and their technological advances have trickled down to less expensive amplifiers. Similarly, the higher performance of modern digital-to-analog converters (DACs) means that the ultrasonic noise generated from noise shaping is much lower than in the past

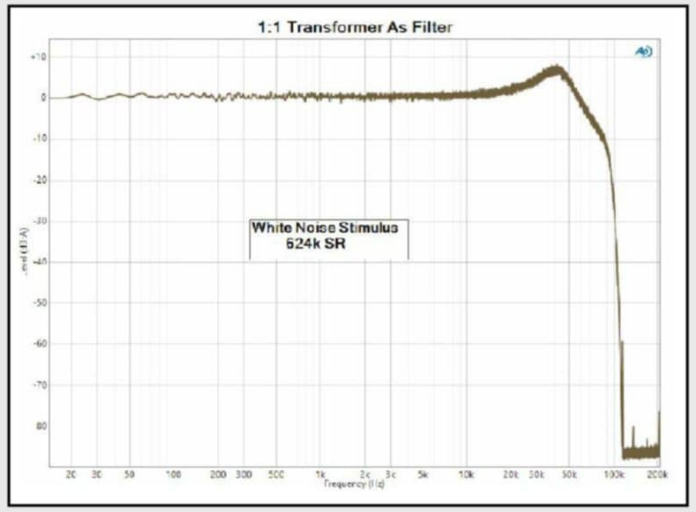


Figure A: Frequency response of the improvised input filter, taken with a white noise stimulus. The increased rolloff after 100kHz is from the white noise band-limiting, not the transformer filter.

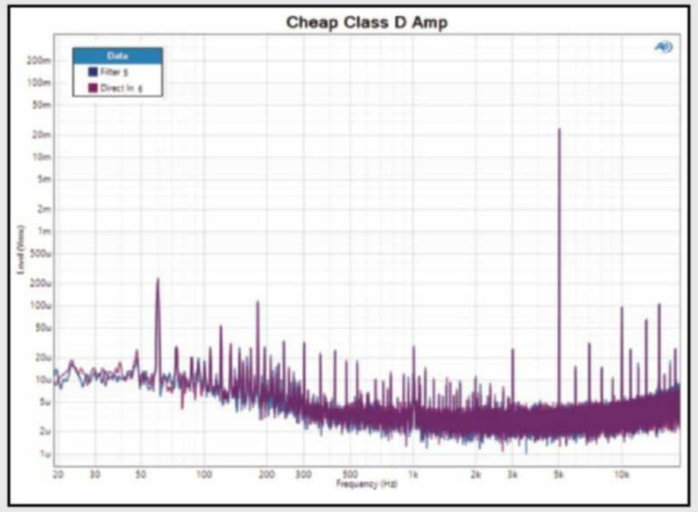


Figure B: A comparison of the noise and distortion spectrum of a modern inexpensive Class-D amplifier with and without the bandlimiting filter.

These advances raise the question of the need for pre-filtering when measuring Class-D amplifiers in 2022. To test this, I ran a few measurements on Class-D power amplifiers with and without pre-filtering using an APx-525 analyzer. Unfortunately, I did not have the external AP filters, such as the AUX-0025 mentioned in the article, so had to do some improvising. I had some Jensen JT11P-1 1:1 input transformers on hand, which have a somewhat gentler low-pass rolloff (12dB/octave) than the more complex passive filters used in the AUX series filters when the secondary is loaded by the analyzer's 100kΩ input impedance and a paralleled 2n2 capacitor (**Figure A**). Nonetheless, they do offer significant reduction in the level of ultrasonic signals, especially at higher frequencies where slew distortion could occur. The 100kΩ /2n2 load is not optimal as can be seen by a slight peaking in the response just before rolloff, but that can be corrected with an RC network across the transformer secondary. And of course, unlike AP's accessory, the transformer can only be used at low signal levels (2V or less), but it's at lower ranges where the band-limiting is most critical.

Obviously, I cannot test every Class-D amplifier out there, but I had four in-house to use as guinea pigs, specifically an nCore, a Purifi Audio, and a GaN Systems representing the high-end high performance sector, and an inexpensive (less than $50) Dayton stereo 50WPC board.

To put some baseline numbers on this, the highest ultrasonic noise for each of the units occurred at the amplifier switching frequency, which was typically 300kHz to 500kHz at a typical level of 0.5V. This corresponds to slew rates in the 2 V/μs to 3V/μs range, which could indeed cause a problem with the older AP System Two analyzer at its most sensitive ranges (40mV to 300mV). Likewise, these out-of-band signals could prevent the APx analyzer from switching to its highest sensitivity range, affecting the measured noise floor.

For the purpose of illustration, I measured the output frequency spectrum of each of the Class-D amplifiers on my lab bench with and without the Jensen input transformers for band-limiting, with the amplifiers set for 25mV out at 5kHz so that the analyzer was being tested in its most sensitive range. The comparative results for each of them were similar. The Dayton amp had the most output noise and distortion, so in the interests of space, I present its results as a proxy (**Figure B**). There is essentially no difference between the measurement made with the analyzer connected directly to the amplifiers and the same measurement but with bandwidth limiting by the transformer.

Note that this is a VERY limited sample set, but does illustrate that there's been measurable progress in both lab instruments and amplifiers in the years since Hofer's paper. That said, I would still caution that before using an analyzer "bareback." One should follow Hofer's advice and use a wideband oscilloscope to capture the noise and determine if there could be a significant measurement problem before trusting the measurement results. As a preventative measure, a filter such as the AUX-0025 is not inexpensive, but makes the measurements significantly more foolproof.

About the Author

Stuart Yaniger has been designing and building audio equipment for nearly half a century, and currently runs a technology consulting agency in western New York. His professional research interests have spanned theoretical physics, electronics, chemistry, spectroscopy, aerospace, biology, and sensory science. One day, he will figure out what he would like to be when he grows up.

The Cranberry Companion

A High-Performance Preamplifier Using Modular Construction

By
Bruce Heran

Photo 1: The preamplifier matches the Cranberry amplifier I had previously built, and thus the name Cranberry Companion. I liked the combination so much that I refurbished and upgraded a turntable to match.

To complement his own "Cranberry" stereo vacuum tube amplifier, our DIY enthusiast decided to build a preamplifier inspired by a modular approach and using solid-state components exclusively. The actual project features a phono, line, and headphone preamplifier stages.

Six milliseconds doesn't sound like much, but in the realm of solid-state electronics it is an eternity. More on that and how it related to portions of the current project later. This project assumes the builder will have some knowledge and skill regarding construction of audio equipment. I would not recommend it for a first project.

I like DIY projects and will often design and build something that I really don't need just for the learning experience. This project is one such build and it ended up being the perfect "companion" to a stereo vacuum tube amplifier that I had previously built (**Photo 1**).

In all honesty, I didn't need an additional preamplifier as I already have several, but the overall concept was appealing and I had all the components to create it. It is a outgrowth of my build in modular design process. I tend to do this with all builds as it lends itself well to the tube-based projects I normally build. We use modules all the time, but just don't recognize them as such. I would submit that in a high-end audio system the power amplifiers, preamplifiers, speakers, power filters, and source components are effectively modules. Since none of us get excited by using those modules, then why should we look down on internal modules used as components of a device? There are many of them and they can be quite useful in building projects. Some are standalone units and others better described as actual components.

In this project I use both types. The digital source selector and volume control is a standalone unit and the active circuitry uses integrated circuits (ICs), which if you think about it are generic gain modules. Using modules makes you think about what you want or need to accomplish in a logical manner. If you want voltage gain that is one thing. If you need power gain, that is another. Indeed if you think about what functions you require, it will guide you to a solution of what sort of modules to use.

Nearly all the time I design my various projects in a block form. Thus, using modules comes naturally. I know of other folks that think in terms of the total function of a device and design in what I would call a linear fashion. They start at the input and keep adding things until they get to the end. That is a viable method, but I prefer the modular approach.

The actual project I will use to demonstrate this is a combined phonograph, line stage, and headphone preamplifier. It uses an off-the-shelf-remote-controlled digital source selector and volume control with a digital display. All this device requires is power. This particular device mates well with a companion case that is available (**Photo 2**). It really makes the build easier. I am one of those folks who can't drill two holes in a straight line (think about that). Anything with about 30 holes is going to be a real challenge with the end product demonstrating that problem.

During the design process, I determined that I would use solid-state components in this project. Most of the time I use tubes so this is a bit of departure from the norm. I decided that I would need some gain, an equalized phonograph preamplifier section, and it was desirable to include a headphone amplifier.

The actual design of the individual stages is pretty generic. A line stage with modest gain using an IC is almost universal. The same is true for the phonograph preamplifier section. I did however tweak the equalization from the calculated values to obtain slightly better test results. There are several possibilities for the headphone amplifier and indeed if the intended headphones are high sensitivity, the line stage could power them directly. It can provide a few milliwatts all by itself. I chose to use an IC headphone module that is widely available to handle the chore as it can deliver a significant amount of power output. You can mix and match the modules to suit your need. The only one actually needed for a preamplifier is the line stage.

Details

Providing suitable power for each of the devices is, in my opinion, key to a good project. In this case there are three types needed: 12VAC, 12VDC, and a symmetrical + and – 12VDC. In earlier projects, I found that one of the easiest ways to get really clean symmetrical DC was to use off-the-shelf modules. You certainly can use a conventional power supply, but the component count goes way up and often the results are not as good. I used a module that will take any AC voltage from around 90 to 250 as input and will deliver 12VDC. These come in many sizes and the ones designated for medical use with the output isolated from the input are the best. **Photo 3** shows 60W, 30W, 15W, and 10W versions of them. They are inexpensive and generally in the $20 to $30 USD range.

To obtain the symmetrical output, I used a second similar module that will take anything from about 8V to 18V DC and provide the regulated symmetrical voltages I want. It is also shown in Photo 3 and costs less than $20 USD as well. Regardless of the ones used, they need to have the output isolated from the input. I am wary of knock-off ones from other brands. They are cheaper

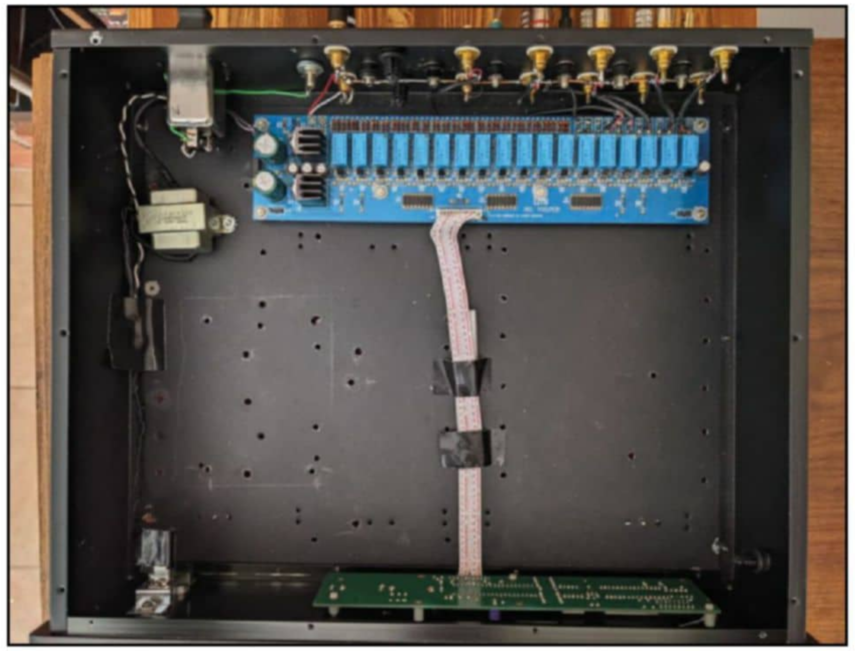


Photo 2: This project uses an off-the-shelf-remote-controlled digital source selector and volume control with a digital display. All this device requires is power.

Photo 3: I used a module that will take any AC voltage from around 90 to 250 as input and will deliver 12VDC. These come in many sizes including the 60W, 30W, 15W, and 10W versions shown here.

and probably work. I like to use known, trusted components for power components, particularly when attached to the AC mains. I like to add a few filter components on the module's outputs as a precaution against high-frequency noise, but they may be optional. In combination, the power modules provide really clean, regulated power. The ICs tend to be immune to some power supply hum and noise so the precautions might not make that much of a difference.

At this point, it would seem to be simple to just hook everything together and it might work. I chose a more sophisticated option. I wanted to have output muting (more on this later) and muting of the line output when headphones were used. To accomplish these functions, some sort of switching would be needed. Relays are a natural for this. The ones I chose run on 12VDC provided by the AC to DC module. I used a front panel switch to turn the preamplifier on and off. This was the simplest way to accomplish the desired action. All of this is well and good until you put it all together. Each part worked fine by itself, but I ran into an unexpected issue with the digital control.

This is the point when the project became a learning experience. All the features of the digital control are via relays. The problem was one of timing. Relays take a finite time to change states. Often in the area of 6ms. As I stated at the beginning of the article, it doesn't seem like much, but in electronics, it is a huge amount of time. What happened was that the normal state of the muting relay in the digital control was open. Thus the output was not muted until the relay functioned. This matters only on the default input number 2. If your signal source is already active, you end up with what I would call a loud "burp" of sound. Certainly undesirable. My solution was to use the normally open set of contacts on another relay to disconnect the output. It was activated with an IC-based (LM555) delay timer. This provided about 4 seconds of delay before the active circuitry was allowed to feed the output terminals. The actual problem is possibly unique to the digital module. It would be possible to short the output to ground as well. Either way works. The way I chose made it easier to enable the headphone only option. Another simple solution is to either not use that input or only attach slower start up source devices to that channel. I used the same digital module in my main stereo system but have a tube-based phonograph preamplifier attached to it. It doesn't make any sound for several seconds and thus avoids the problem.

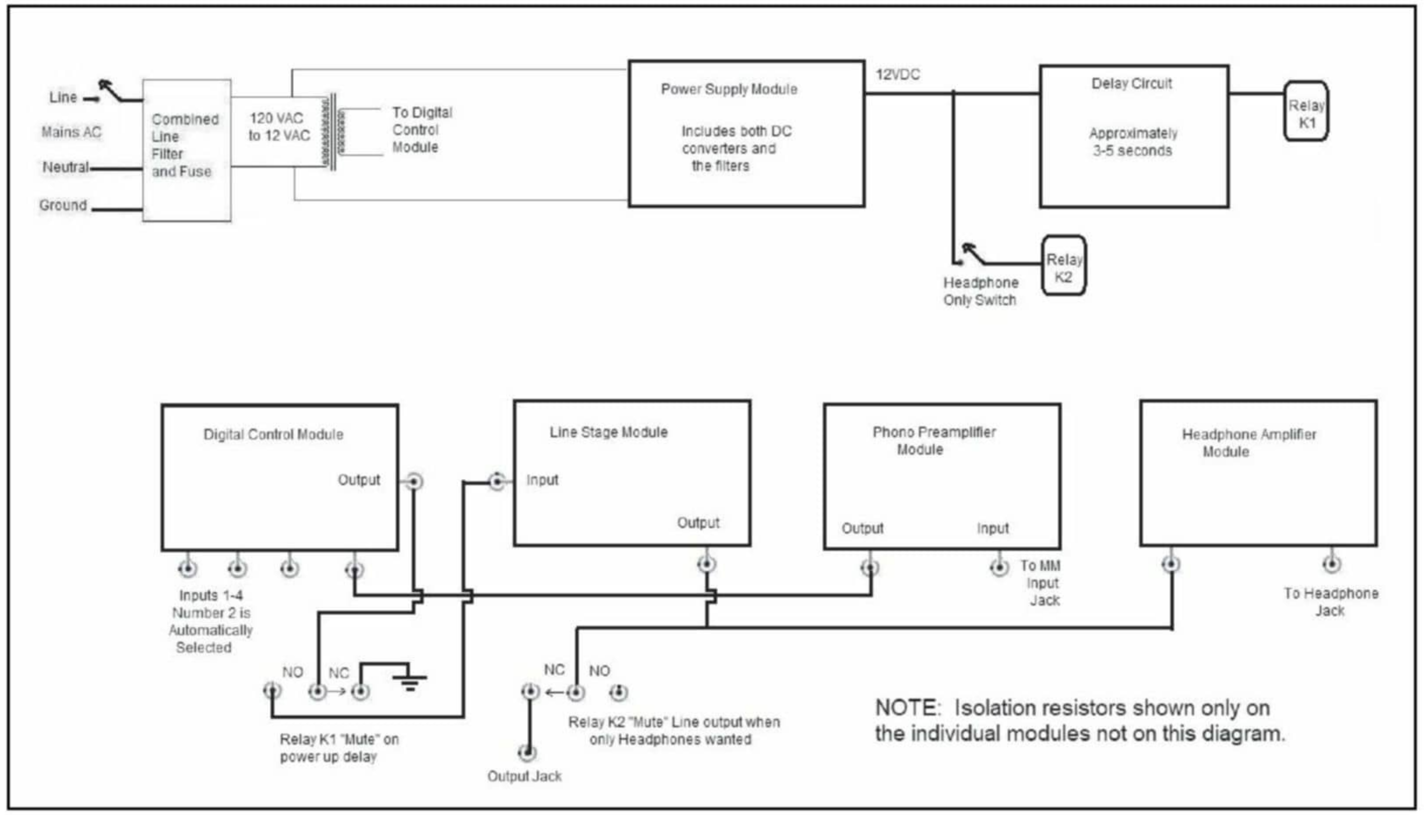


Figure 1: I separated the individual module schematics so that it would be easier to pick and choose which ones are desired and to avoid a really complicated unified schematic. This block diagram shows the interconnections.

Construction

DIYers come in all types. Some insist that everything must be point to point. Tongue in cheek, I call some of them spaghetti builders. I am sure you have seen some examples. Wires and components going in all directions. It often resembles a bird's nest. Many are new to the DIY field and have not yet figured out that it is really hard to troubleshoot spaghetti wiring and even if it works it can result in signals, hum, and noise coupling between various sections.

My experience has been that using terminal strips, turret boards, and perforated boards is far better for beginners. In my opinion, printed circuit boards (PCBs) are better for more advanced builders. Any method can be used in good designs in the hands of a skilled builder. The extreme iteration is having the controls, input, and output jacks and sometimes even the power connections on the board. As a commercial design this is fine, but for DIY it seems to nearly always cause problems. These usually occur in the form of compromises in component and control locations. I have found that few builders can get their first try at the PCB design to work as expected.

A third type is those builders who use a combination of forms, selecting what works best for what the design needs. I fall into that group. That leads me to the project at hand. This is an integrated preamplifier. It includes a number of features I have found useful in the past. They are a line stage for a minimal gain, a phonograph preamplifier that can be customized to both high and low output cartridges as well as NAB tape heads, a headphone amplifier and possibly most importantly to many folks a fully remote digital control assembly. All in a modular type of arrangement. This also brings up the thought that many DIYers consider using modules horrible. The "if I didn't make it, then it is no good" attitude is encountered from time to time. It overlooks the obvious as many components (e.g., ICs and power supplies) can already match the definition of modular.

I don't intend to criticize anyone but I am presenting a design based on the use of modules. Modules such as those in the preamplifier are not plug and play. They need to be utilized and properly interfaced to achieve the desired results. I separated the individual module schematics so that it would be easier to pick and choose which ones are desired and to avoid a really complicated unified schematic. A block diagram is included to show the interconnections. Please note that components like filter capacitors and isolation resistors that are on the individual module boards are not shown in the block diagram (**Figure 1**). They are necessary so don't omit them. The block diagram does include the line filter (**Figure 2**) and the start-up delay circuit (**Figure 3**), although these components are not discussed further in the article. The gold covers over some assemblies are mostly cosmetic and are a carry over from a tube-based preamplifier with a much higher circuitry impedance and in that build the covers provided a small amount of shielding.

Digital Control

I first decided to use an off-the-shelf digital remotely operated "volume control" that had input switching. There are several other digital and analog controls available. Some rather inexpensive. I ordered a case that matched it for ease of construction. Be aware that the module I chose is about $130 USD and the case (that comes with all the hardware, power switch, and jacks) is about $75. The issue is that it is heavy and the shipping is quite high. Depending on the destination, it can easily be more than the price. For me, it was a huge time saver and really enhanced the build. The digital assembly requires only a source of 12VAC for operation. Any small 12VAC transformer will be quite satisfactory. If you use a different control be aware of the possibility for ground loops.

The digital control assembly that I chose separates the power and signal grounds and avoids this problem. Others I have tried do not and the result is ground noise contamination that is hard to eliminate. If the remote control features are not desired, a more simple design using a variable resistor and multi section switch would do the job. It would be significantly less costly as well.

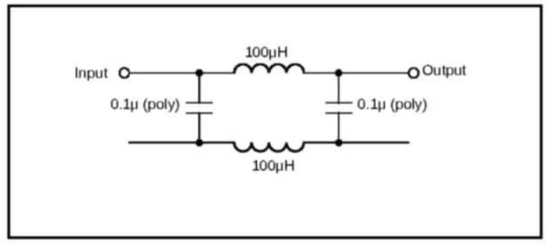


Figure 2: The line filter and fuse (120VAC to 12VAC) connects to the digital control module.

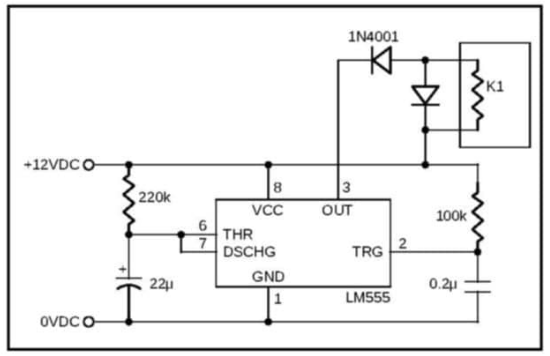


Figure 3: The start-up delay circuit is an important component that helps bridge the gap between the power supply and the relay.

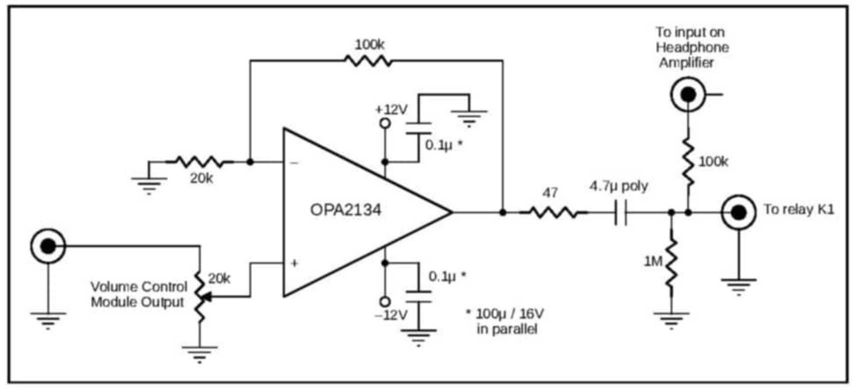


Figure 4: The output from the line stage goes to the muting relay and headphone amplifier.

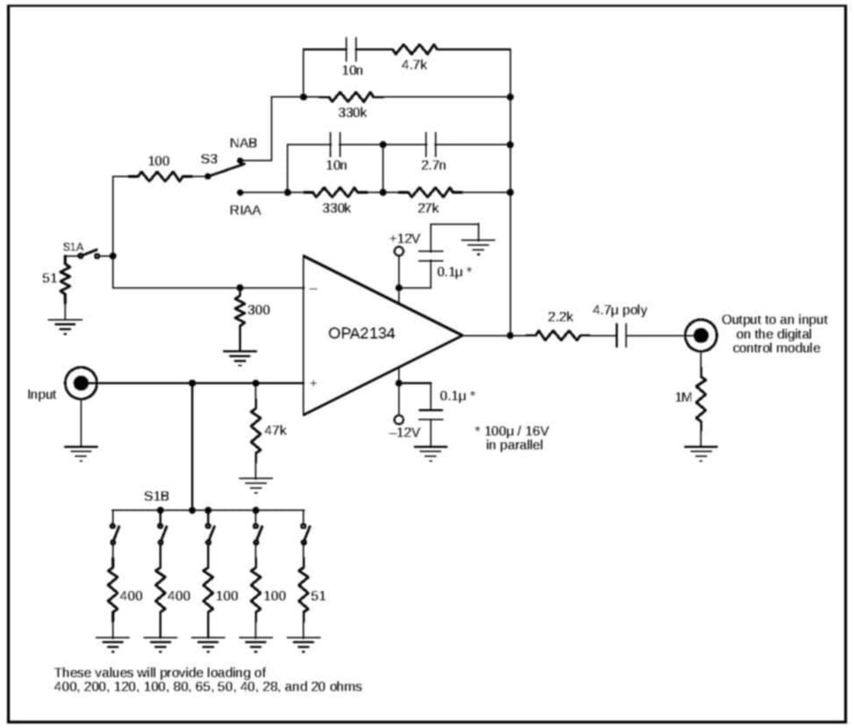


Figure 5: The phonograph preamplifier section uses an OPA2134. It too is only coupled at the output.

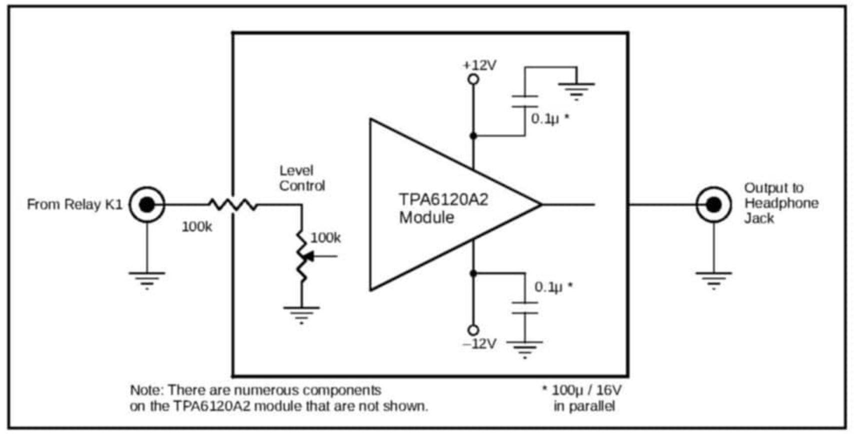


Figure 6: The headphone module gets input from the line stage directly and is always on.

Line Stage

The line stage is based on a OPA2134 IC. These ICs are commonly available from many different sources. Mine were made by Burr-Brown. They are inexpensive and perform well. The module is set for a gain of 5. Other values can be chosen to suit your needs. To do that, it is necessary to alter the ratio of the feedback resistor or the one from the non-inverting input to ground. Given a choice I would alter the feedback one. The gain is roughly the ratio of the two resistors. It is a typical non-inverting stage operated at full bandwidth. I capacitor couple the output as it frequently is not exactly at zero VDC. Yes, it is possible to put capacitors in the inputs between the resistors and ground to prevent any DC gain, but my experience has shown that the fewer components in the input signal path the better and an output capacitor is still likely to be needed. It will avoid any DC voltage reaching the next stage.

Once in a while you will find an IC that misbehaves and "locks up." The easiest thing to do is replace it. I always get a few extra ICs just in case. Buy them from trusted sources as there are many fakes on the Internet. Curiously, any high-quality dual-channel IC with the same pin arrangement (most are) will work in the circuit. Folks who use similar circuitry will often select others that they like better. Some are lower noise, some lower distortion and some higher bandwidth. I like the OPA2134s because they are easy to get, perform well and trouble free. The output from the line stage goes to the muting relay and headphone amplifier (**Figure 4**).

Phonograph Module

The phonograph preamplifier section also uses an OPA2134. Many folks might argue that one IC is not really suitable for this application and that at least two are needed. I have found that in this design a single one works just fine. It has sufficient gain and bandwidth to handle the frequencies needed. The response of the stage is altered by an equalization network in the feed back loop. It sets the overall gain and equalization needed to comply with the RIAA standard. It can be set to match both high and low output cartridges and high or low impedance ones. It can be set for a general-purpose tape head preamplifier with gain and response that matches the NAB curve fairly well with many tape heads at the 7.5 inches per second (ips) speed.

The stage is adjustable and a myriad of combinations are possible. The design of the stage is similar to the line stage and the same considerations apply. It too is only coupled at the output (**Figure 5**).

The output is routed to an input on the digital control assembly. All the other inputs on the control assembly are used at line input levels. The typical gain as shown in the schematic is approximately 40dB with an alternate at 60dB. Adjustments to the feedback loop can achieve tailored gain levels but with only a single IC the gain really needs to be limited to a maximum of about 60dB. The 300Ω and 51Ω resistors on the inverting input terminals provide these gain settings. Values in between them can be used to set other gain levels. Gain above the 60dB level would encroach on limitations of the device and is not recommended. The NAB setting is for 60dB.

Headphone Amplifier

The headphone amplifier is a TI6120AP IC. The one I used was mounted on a mini PCB that already included the components needed for operation. There are several sources for similar ones and I would encourage getting it this way. They are quite inexpensive. Some cost less than $5 USD. Otherwise, you will need to be skilled at surface-mounted IC component soldering and have to obtain a significant number of standalone components. This is an example of using a module to perform a function, cheaper, easier, and likely better than can be done with doing things the hard way.

A word of caution is in order as there are some defective versions online. The problem is that one of the electrolytic filter capacitors is installed backward and will prevent operation of the module. A continuity check of the connections from the power input terminals, capacitor terminals, and the IC will identify if there is a problem. The easiest solution is to remove the capacitors and use external ones.

This module gets input from the line stage directly and is always on (**Figure 6**). There is an isolation resistor of 100kΩ between it and the line output jacks. Activation of the "mute" switch disconnects the signal from the output jacks but still allows it to feed the headphone amplifier. I included a pair of level controls on the headphone amplifier PCB so that common volume levels at the digital assembly output match well with reasonable headphone output levels. Depending on the impedance of your headphones up to nearly a watt of output power is available. It will operate with impedances as low as 8Ω.

Power Supply

Each of the modules in this preamplifier is decoupled from the power sources and each other.

Parts Information

All resistors are either metal or carbon film and can be ¼, ½, or larger in dissipation. 5% tolerance is fine, but with inexpensive 1% ones now available now most were 1%. Do not use carbon composition resistors as they tend to be quite noisy.

All capacitors were rated 25V or higher with the exception of the X2. It needs to be rated for the value of the AC used. Typically the voltage rating is 275VAC.

The rectifiers used to suppress spikes on the one relay are generic 1N4001. Nearly any rectifier will be fine.

The relays are a generic type HK19F-12V. They are quite inexpensive and many other types can be used.

The AC to DC converter module is a Meanwell MPM-10-12. Most large parts suppliers carry them.

The DC to DC converter module is a Meanwell DCW03-12. Most large parts suppliers carry them.

The digital control assembly is a Lite V02. A web search will locate sources. Prices vary considerably on this assembly and you need to check each one for hidden shipping costs. A plastic cased remote comes with it standard. A higher quality metal cased remote is also available for about $40US.

The case I used is a Lite A28C. This version has the window for the digital display. All others in the A28 series do not. Typical cost for the case is about $70. Beware that as a rather heavy item, shipping can easily exceed that value. It cost me more than $100 to get it shipped. Well worth it in my case.

The 12VAC power transformer was a generic one. Acquire locally to match your AC mains voltage.

The integrated circuits (OPA2134 and LM555) and headphone amplifier module (TPA6120A) are either local or sourced online. Beware of fakes and get them from your chosen reliable sources.

The small module size circuit boards shown in the photos were from Marlin Jones and are recently discontinued. They are not needed for the build and others will be fine.

The power entry, line filter, fuse, and main power switch should be chosen to suit your mains voltage configuration. There are many choices and sources. They must handle at least one ampere (1A). Power on the indicators are optional and should be designed to operate on one of the DC voltages as a check that the circuitry is functional.

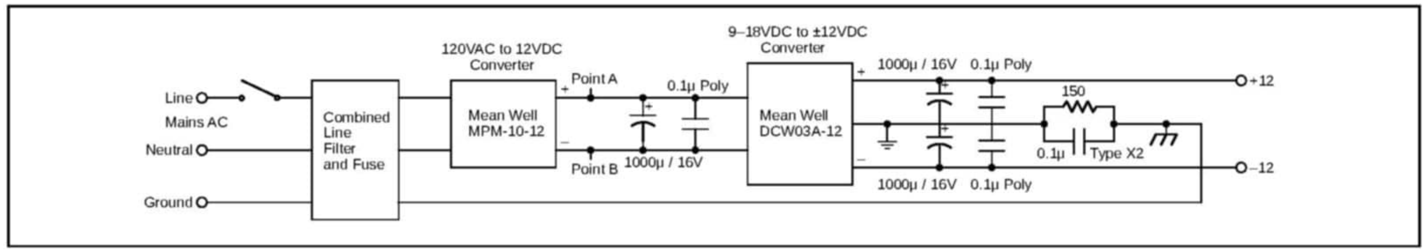


Figure 7: Each of the modules in this preamplifier is decoupled from the power sources and each other. To get really clean DC power, I use a modular configuration. I used power supply modules to obtain the necessary DC power.

To get really clean DC power, I used a modular configuration. As mentioned earlier, I used power supply modules to obtain the necessary DC power (**Figure 7**). A Meanwell MPM-10-12 AC to DC module is used to obtain a clean 12VDC at 0.85A. This is frequently used in medical equipment for this purpose. It is clean, stable, low noise and the output is isolated by thousands of volts from the inputs.

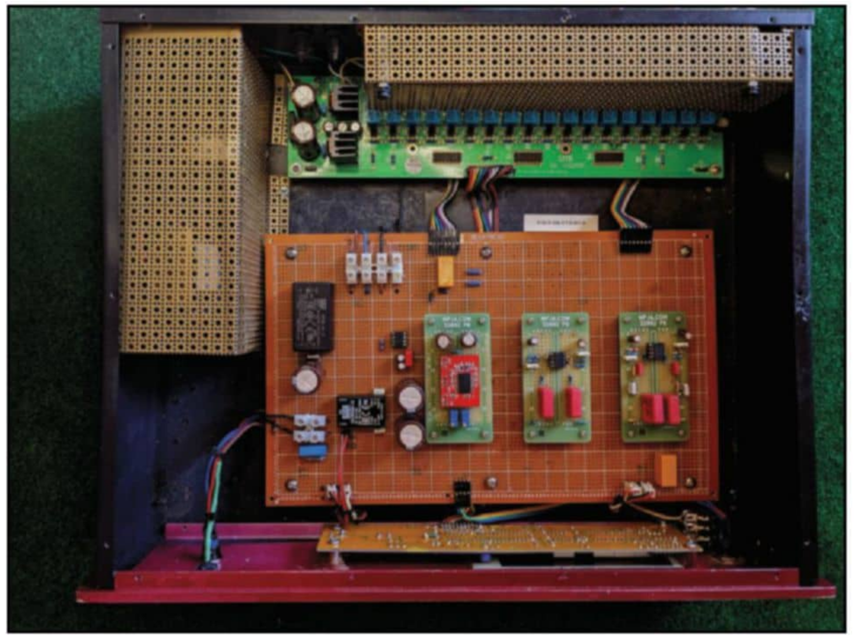


Photo 4: As in any audio project care needs to be taken when placing and routing wires and components. In a departure from many of my previous projects, I made extensive use of ribbon cables in this one.

Photo 5: Using the preamplifier was as simple as plugging it in and selecting the source and volume desired. The front of the "Cranberry Companion" (a) and the rear of it (b) are shown here.

It will operate on any AC voltage from about 90 to 250. The output powers the relays and feeds another module. This module is a Meanwell DCW03A-12. It takes any DC voltage input from about 8 to 18 and converts it to a regulated symmetrical ±12VDC. It is quite clean, stable, and low noise. It also is isolated between the input and output.

I have used these modules in numerous projects with excellent results. They simplify designs and are low cost. Other similar modules by this company are quite useful and I have used some in tube preamplifiers for heater power. Just be sure in that case to up size them to about twice the needed current. If you don't, then it can sense the load as a short and shut down. I do take extra steps in the power supply chain by adding filter components to ensure the cleanest current and lowest noise level. It may be overkill, but I really like my projects to be quiet and it is inexpensive and simple to accomplish.

Basic Construction

Construction is pretty straightforward and generally poses no issues for most builders. As in any audio project care needs to be taken when placing and routing wires and components. Signal carrying wires and components should be kept away from ones that carry AC voltages. In a departure from many of my previous projects, I made extensive use of ribbon cables in this one (**Photo 4**). I decided this would be satisfactory as the modules utilize rather low impedances compared to the tube circuits that I frequently use.

Most impedances are in the range of one to two orders of magnitude lower. Even so, I separated the signal conductors apart with ground connectors. These were attached to the signal ground circuit at only one end to avoid ground loops. I used a large gauge ground type of buss and separate power buses in the layout for convenience. There has to be careful attention to avoid ground loops throughout the build.

Each module was connected to the ground at only one point and the main board had only one connection to the control assembly ground. Care was also used to insure that none of the various power sources were connected in ways that would compromise the build. The outputs of each one is isolated from each of the others.

A single point of contact between the signal ground and chassis was made via a type X2 capacitor and parallel resistor. X2 capacitors are rated for use in this manner and other types should not be used and in some cased not allowed by national electrical codes. This is one of two common methods. The alternative is to use a pair of rectifiers in parallel with opposite polarities. Both ways allow the chassis to operate as a safety barrier from internal component failure and as an electromagnetic interference (EMI) shield for the circuitry. Both designs require the use of three wire mains AC connections.

Testing and Initial Setup

You can check out the sections individually or do everything all at once. Considering the nature of the voltages involved (except for the few locations of AC mains) and the low cost of components, I tend to favor the all at once method. If everything works as designed, great. If not, it is usually pretty clear where the problem or problems exist. Using common components and easy to check modules simplifies trouble shooting. One area that can cause consternation is the operation of the relays. Miss wiring them can fool you into thinking one or more of the modules is not operating correctly. The only adjustment needed is setting the headphone level controls. Since the digital control has an 80dB range I arbitrarily set it at the halfway point (-40dB). Then with a clean signal from a high resolution player set the headphones to a comfortable level. Depending on the power needs of your headphones your level could be higher or lower.

Use

Using the preamplifier was as simple as plugging it in and selecting the source and volume desired. I designed it around an amplifier that I made a few months ago that was published as a project in *audioXpress* (May 2021). The preamplifier (**Photo 5**) matches the amplifier and thus the name Cranberry Companion. I liked the combination so much that I refurbished and upgraded a turntable to match. It started life as a Pro-ject Debut III, but now has so many modifications and upgrades that it is quite different and much improved over the original.

About the Author

Bruce Heran lives in Sierra Vista, AZ, and is an avid DIYer. He has been involved in electronics for 59 years. He built his first project—a one-transistor amplifier—when he was 13. He has a BS in Biology and studied electrical engineering for two years. Electronics has always been his hobby. In May 2011, he retired from his work as a project manager of a logistics contractor with the Federal government. Now he is co-owner of Oddwatt Audio, an electronics company that specializes in vacuum tube audio components. He is vice president of design and support, and all of the company's products are kits and assembled audio equipment based on his personal designs. He is an avid collector of vinyl records and vintage turntables.

Egg Speaker Amp and System

By
Reinhard Metz

Photo 1: These are my completed Egg speakers, flanked by the CS7s, which are about to be replaced.

Inspired by Tom Perazella's egg-shaped speaker project, the author was motivated to build his own version and explore his take on the design with an active crossover tri-amp approach using a miniDSP 4x10HD system feeding a dedicated six-channel amplifier, including Purifi 425W Class-D units.

It started with Tom Perazella's extraordinary Egg speaker three-part article series featured in the September, October, and November 2018 issues of *audioXpress*. My trusty KEF-designed CS7 kit-built speakers had performed admirably for many years, but I had been thinking for some time about building a more modern system. The Eggs and their comprehensive build info clinched it—this is what I was going to build! From there, one thing led to another.

Egg Speaker Design and Construction

First, I made a deviation in the woofer section. I did not want as much footprint as Tom's pyramids presented. That fit right in with the learning I had hoped for in this project. I've never designed a speaker and frankly have been intimidated by some of the design ingredients, if you are going to go about it in a serious way. Vance Dickason's *Loudspeaker Design Cookbook* had sat on my shelves forever and now it was time to read and understand and apply it. I would design my own cabinet to meet a set of criteria, still using the same Dayton Audio RSS315HO44 12" woofers from the Egg design. I won't dwell on the midrange and tweeter egg section, having pretty much followed Tom's construction descriptions and using the same Satori TW29BN-B beryllium dome tweeters and MR16P-4 midrange drivers. **Photo 1** shows the finished product in place, flanked by the CS7s, which were about to be replaced.

Woofer Section Design

The design process began with a combination of a rough footprint, desired height (with the egg section still designed to sit on top), and a mental image of shape, admittedly somewhat influenced by the venerable Waveform Mach 17s. The next step was to apply some of the design theory to determine a required volume. It's an iterative process given the footprint and size constraints vs. the resultant performance. This was accomplished using the published speaker parameters of F_s, Q_{ts}, and V_{as} to calculate Q_{tc}, V_c, and F_c for a range of V_c ft^3 values from 1 to 5 (**Table 1**).

From here, it was a matter of deciding the trade-off of low frequency cutoff vs. tightness of bass desired, leading to a compromise volume around 4.5ft^3. The shape I had in mind was for an upper and lower section of a relatively rectangular and columnar approach, the lower section slightly canted upward, and the upper facing straight into the room. I also wanted complimentary slight angles to break up what would otherwise be straight sides.

Initially, I built some 3-D models, but somehow I wasn't quite satisfied with the visualization of the actual look. So instead, I built some full-scale models with duct-taped cardboard, which turned out to be quite effective. It also allowed my wife to add her approval to what would eventually be appearing in the living room! **Figure 1** shows the dimensional result after playing with variations of the dimensions in a spreadsheet that calculated volume, and until the look spoke to me on paper and with the cardboard models, and the volume matched the Vc ft^3 calculation. For the dimensions shown and built, the internal volume is 4.29ft^3. The row highlighted in yellow in Table 1 shows the Q_{tc} at 0.45 and the F_c of about 26Hz that result from this volume. That should provide a decent bass extension with reasonably tight response.

That said, I was now faced with the challenge of cutting wood to meet the design plan. If you've ever built a wood project with multiple odd angle intersections, you know that determining the cutting angles can be daunting. Fortunately, I didn't need to haul out any dusty math books or make my brain hurt too badly! There are several excellent online angle calculators for exactly this kind of wood cutting, one of which did me great service. Still, laying out those angles on the many pieces of wood and setting the saw appropriately over and over still left plenty to do and several trials and errors to get it right.

Five pieces constitute the bottom section, and five more for the top. I doubled the thickness of the front two pieces to facilitate routing for speaker mounting the drivers. All the pieces were joined with biscuits and wood glue in multiple clamping operations. For ease of clamping and alignment, the top and bottom sections were assembled separately, (**Photo 2a**) then

RSS315HO-44 Specs	
Impedance	4+4Ω
R_e	6.5Ω
L_e	3.17mH @ 1kHz
F_s	21.5Hz
Q_{ms}	3.7
Q_{es}	0.41
Q_{ts}	0.37
M_{ms}	327.7
C_{ms}	0.17 mm/N
S_d	507.1 cm²
V_d	720.6 cm³
B_L	26.5 Tm
V_{as}	61.3 ltr
X_{max}	14mm
VC Diameter	65.5mm
SPL	84.6dB @ 2.83V/1m
RMS Power Handling	700W
Usable Frequency Range (Hz)	20-500 Hz
Effective Dia Sd	10"

$Qtc=Q_{ts}*(V_{as}/V_c)+1)**.5$

$V_c=V_{as}/[(Q_{tc}/Q_{ts})2-1]$

$F_c=F_s*(V_{as}/V_c+1)**.5$

$F_c=Q_{tc}*F_s/Q_{ts}$

Q_{tc}	V_c ft^3	F_c	V_c ft^3	Q_{tc}
0.4	12.82	23.24	1	0.66
0.45	4.52	26.15	1.5	0.58
0.5	2.62	29.05	2	0.53
0.55	1.79	31.96	2.5	0.51
0.6	1.33	34.86	3	0.49
0.65	1.04	37.77	3.5	0.47
0.7	0.84	40.68	4	0.46
0.75	0.70	43.58	4.5	0.45
0.8	0.59	46.49	5	0.44
0.454	4.29	26.37	4.29	0.454

Table 1: The design process began with a combination of a rough footprint, desired height (with the egg section still designed to sit on top), and a mental image of shape, designed using published speaker parameters.

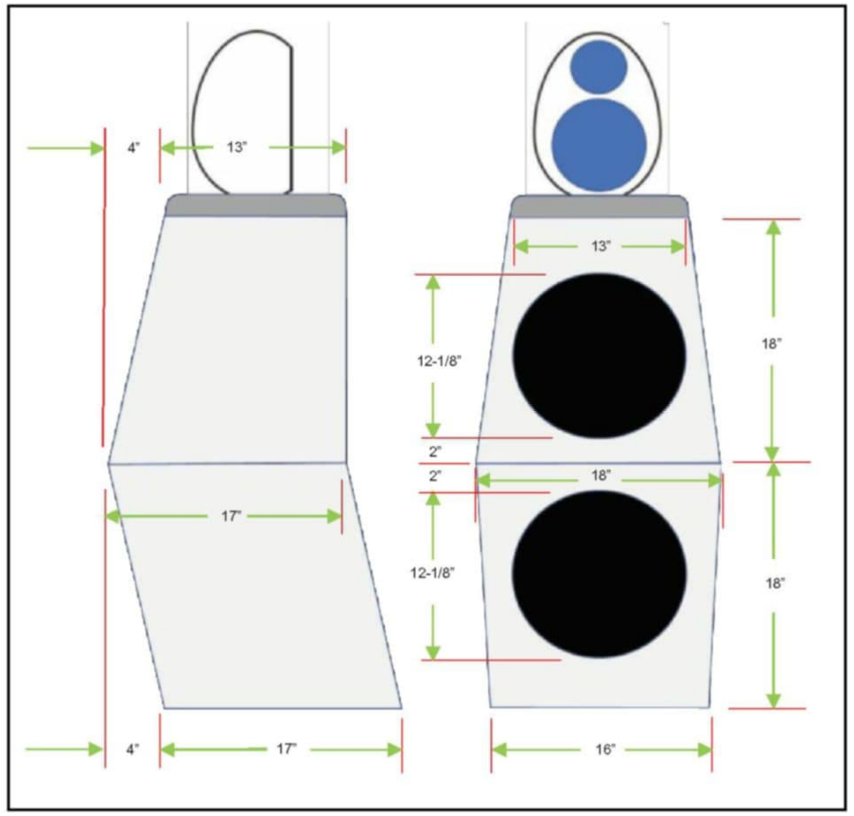


Figure 1: I designed the dimensional result shown here after after playing with variations of the dimensions in a spreadsheet that calculated volume, and until the look spoke to me.

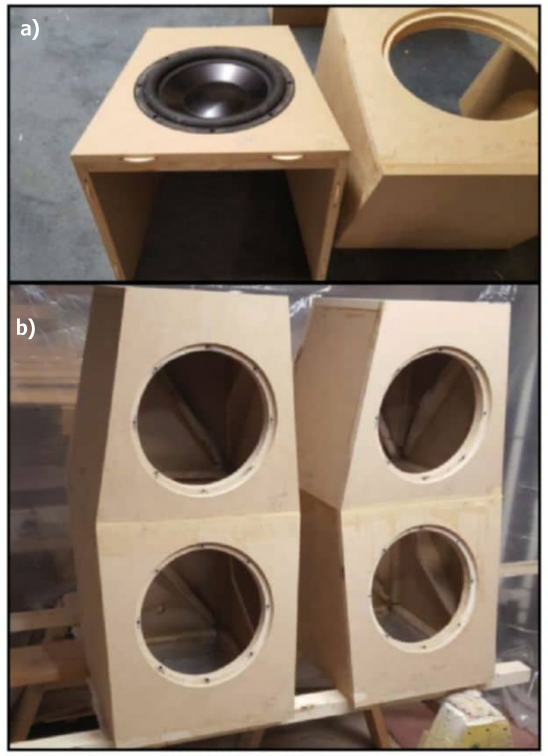


Photo 2: For ease of clamping and alignment, the top and bottom sections were assembled separately (a) then joining top and bottom together until I had the rough completed assembly (b).

Photo 3: Then, I reinforced the front boards with glued in braces.

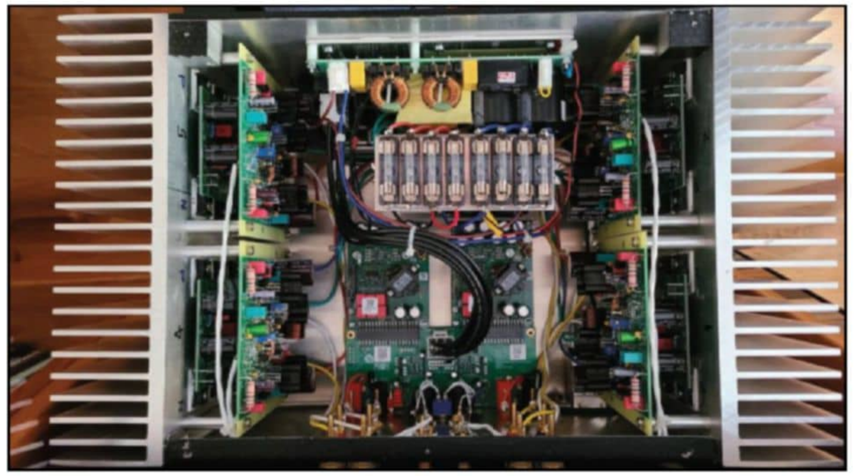


Photo 4: I used an evaluation kit that consists of a pair of the Purifi amp modules along with a shared common low voltage power supply and gain sections—all in a remarkably small footprint.

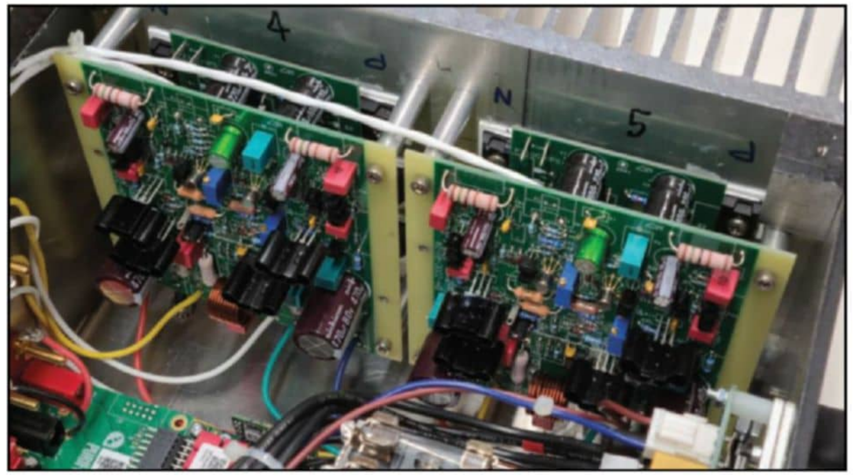


Photo 5: While physically targeted for the original Hafler amp cases, the form factor was ideal for the enclosure I was envisioning.

joining top and bottom together. Rough completed assembly (**Photo 2b**). Finally, all but the front boards were re-enforced with glued in braces (**Photo 3**).

Sanding the board intersections followed by router ½″ quarter rounding and lots of finish sanding brought the process up to the paint stage. I preceded the white paint finish with a coat of sprayable polyester filler, which eliminated any residual texture of the MDF and prevented any differential absorption of the finish coat. A fine sanding produced the desired final uniform and matte finish look.

Crossover and Amplification

The two crossover frequencies chosen were 200Hz and 2500Hz. I began with a passive crossover network based on Tom's article and my existing 250W amplifiers. The results were not as impressive as I had hoped. As the passives have a number of limitations and are expensive and painful to iterate changes, I quickly concluded that I would for the first time try the active crossover tri-amp route. The first attempt was with a three-channel analog filter with individual gains and two crossover frequencies, set by plug-in filter components. This didn't yield an overall flat enough response or satisfying sonic performance either for even my not-so-golden ears.

Time to take one more step into modernization with a DSP crossover. The one I chose as a compromise of features and cost is the miniDSP 4x10HD. It has a substantial set of direct filter frequency and gain adjustment capabilities as well as multiple byquad IIR filter sections that can be fed parameters from several analysis programs. For initial tests, I used three separate amps I had lying around. From the speaker efficiency data and some initial mid-band measurements, referencing to the woofers (they being the least efficient), the midranges needed an input level of -13dB and the tweeters -14.5dB. Programming those gain factors and the crossover frequencies into the miniDSP 4x10HD, I was finally making progress toward acoustic results I liked and that started to measure well.

Amplifier Decisions

Given the above driver input levels and assuming, say 1000W for the woofer section, the midrange requires only 46W and the tweeter 35W to simultaneously reach the maximum output level. Clearly the pile of equal power mono-blocks was not optimal. So, the goal became to also build a dedicated six-channel amplifier, with a pair of higher-power channels for the woofers and identical four lower power amps for the mids and tweeters. Additionally, I wanted a reasonable size and weight package, with the amplifier performance best matched to each driver type.

Amplifier Design

Nowadays, the necessary high power for the woofers is most easily achieved with Class-D amplifiers, with the side benefit of substantial weight reduction. There are a number of Class-D designs available that also have excellent frequency response. I chose a pair of Purifi Audio 425W Class-D amps. Although the RSS315HO44 12″ bass drivers are rated for 700W each, so theoretically capable of accepting up to 1400W each channel, I decided 425W would be enough, a good compromise between SPL and minimized footprint, weight, and heat for the bass channels.

These modules were discussed in an Industry News item on the audioxpress.com website, "Purifi Audio Promises to Reduce Distortion in Speakers and Amplifiers," in April 2019, and reviewed by Ward Maas in the July 2020 issue of *audioXpress.* The modules impressed me back then with their design approach and exceptional performance. Purifi Audio offers an evaluation kit that consists of a pair of the amp modules along with a shared common low voltage power supply and gain sections, all in a remarkably small footprint (**Photo 4** and **Photo 5**). They incorporate a back panel that includes a set of RCA and XLR balanced inputs, as well as standard ¾″ spaced banana jacks for the output.

The goal for the midrange and tweeter channels was to stay with Class AB and to have performance worthy of the outstanding Satori drivers. While I've designed a number of FET-based power amplifiers over the years, I was impressed by Bob Cordell's retrofit design for the venerable Hafler DH-220 and DH-230 amps, as recently featured in the July and August issues of *audioXpress*. PCBs are available for them on eBay (see the Parts List), making the build a whole lot easier than rolling a new design of my own, and their performance and low noise levels put them in the quality class of the driver elements. While physically targeted for the original Hafler amp cases, the form factor of the Cordell DH-230 amps was ideal for the enclosure I was envisioning.

When ordering these on eBay, you will receive the bare PCBs and detailed support for ordering parts, including prepared order sheets for two parts supply houses. Finally, to keep things light, I chose a pair of switching power supplies to complete the assembly (**Photo 6**). The Purifi Class-D amps are powered from a 1200W Hypex SMPS1200A400 power supply, capable of feeding both of the 425W amps to simultaneous continuous full output. The

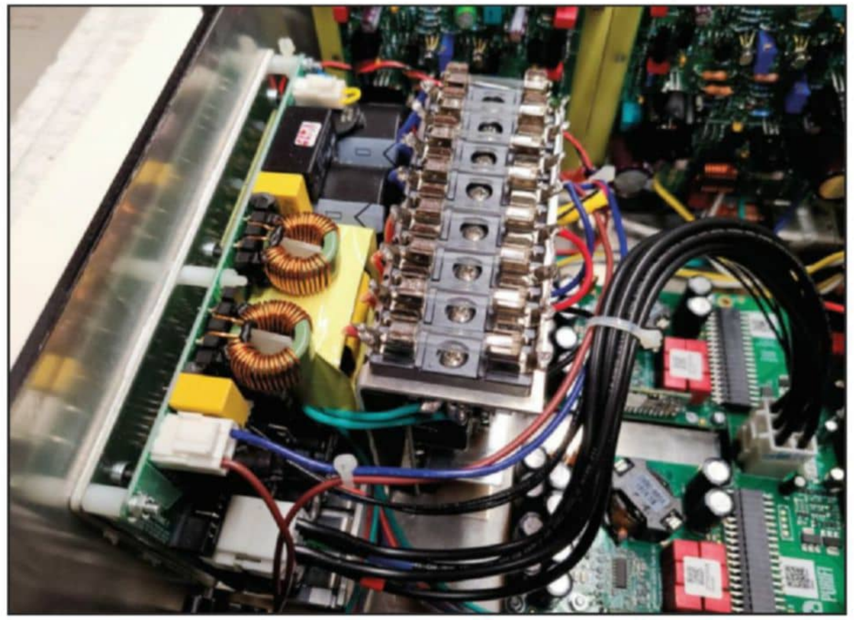


Photo 6: To keep things light, I chose a pair of switching power supplies to complete the assembly.

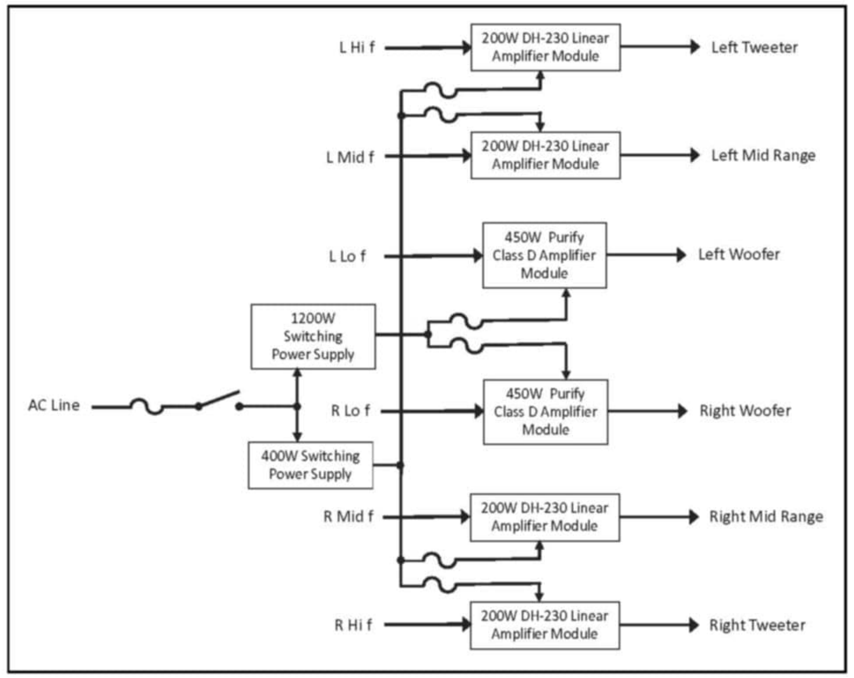


Figure 2: This block diagram shows the six-channel amplifier.

About the Author

Reinhard Metz received his BSEE and MSEE degrees from the Illinois Institute of Technology and University of Illinois/Champaign respectively. He worked for 29 years as an engineer and manager at Bell Laboratories before starting an engineering and legal consulting business. He then joined CookTek LLC. where he specialized in power electronics for commercial induction cooking equipment and became V.P. of engineering. Now retired, he has had the opportunity to return to his life-long audio hobby, including the design and building of FET power amplifiers. He has published 27 papers and articles in the areas of Telecom, audio, and general aviation. Aviation being his other consuming hobby, he flys a four-seat fiberglass high-performance airplane, which he built himself. He has 18 patents and three pending.

four mid and tweeter amps are powered by a 400W Hypex SMPS400A400 power supply. The latter will clearly not enable all four amps to deliver their 200W simultaneously, but in practice the power required to drive the speakers is maximum in the neighborhood of 20W per midrange and 15W per tweeter. These amps will be loafing, but were chosen specifically because their performance is so exceptional, including when operating at low power output!

Figure 2 shows a block diagram of the six-channel amplifier. Each amplifier is separately fused at the DC input side. The physical design is anchored by a pair of 6″ tall Wakefield 510 heatsinks on the sides and a thick aluminum front plate that serves double duty as the heatsink for the 1200W switching power supply (**Photo 7**). The back panel (**Photo 8**) mounts the Purifi I/O, 6 RCA inputs, and six sets of gold binding posts for the speaker output.

Crossover Tuning

Tests and measurements were done with the Dayton OmniMic speaker measurement system. The initial programming parameters I used for the MiniDSP were crossover Linkwitz-Riley fourth-order crossover frequencies of 200Hz and 2500Hz, and attenuation factors of 0dB for the woofers, -13dB for the tweeters and -14.5dB for the midranges. Initial listening and OmniMic scans suggested the need for an additional -6dB filter at 2kHz to suppress a slight nasal intonation as a tuning starting point.

Photo 7: The physical design is anchored by a pair of 6″ tall Wakefield 510 heatsinks on the sides and a thick aluminum front plate that serves double duty as the heatsink for the 1200W switching power supply.

Photo 8: The back panel mounts the Purifi Audio I/O, six RCA inputs, and six sets of gold binding posts for the speaker output.

Project Parts List

Qty	ID	Description	Source	Comments
Speaker				
4	RSS315HO-44	Dayton Audio Bass Driver	Parts Express	
2	MR16P-4	SB Acoustics Satori Mid Range Driver	Madisound	See *audioXpress* September, October, November 2018 issues
2	TW29BN-B	SB Acoustics Satori High Freq. Driver	Madisound	For Egg portion of the construction
20 lb.	Acousta-Stuf Polyfill	Speaker Cabinet Damping Material	Parts Express	
2	NL4MP	Neutrik Speaker Connectors	Parts Express	
2	NL2MP	Neutrik Speaker Connectors	Parts Express	
3×	4x8x3/4″	Sheets of MDF Wood	Home Depot, Menards, ETC.	
Amplifier				
1	345-1962-ND	Wakefield 10WX12″ EXTRUSION	Digi-Key	Cut in half
1	3/16″ Aluminum front panel	89015K276	McMaster-Carr	
1	0.062″ Base Plate and back	89015K189	McMaster-Carr	
2	Handles	2471N1	McMaster-Carr	
	Top cover	Perforated Steel	McMaster-Carr	Bend to shape
1	EVAL1	Dual Class D amplifier kit	Purifi Audio	
1	SMPS1200A400	1200W Switching Power Supply	Hypex	Order with amplifier interconnect cable
4	DH-230C	200W Amplifier Module Kit	eBay, item #223993422898	eBay has the PCBs, they come with comprehensive parts ordering support through DigiKey
1	SMPS400A400	400W Switching Power Supply	Hypex	Order with connectorized cables
12		Banana Jacks		
6		RCA Jacks		
8		Fuse holders		
		Miscellaneous Hardware		
		Miscellaneous Wire		
1		Pilot light		
1		Power Switch		
1		Back Panel fuse and holder		

Further Tuning and Performance

Figure 3 shows the measured frequency at the typical listening position with 1/12th octave smoothing, and the impulse response measured 5′ in front of the speaker. Beyond the crossover filters, frequency response compensation at this point consists of several manually generated IIR filters using the miniDSP plug in application. Notably, very little compensation was required to flatten the high end, and no boost filtering whatsoever was applied at the low end. The flat extension down to around 12Hz or 13Hz is innate, which took me quite by surprise given the driver's F_t around 20Hz. Room location with the speaker a foot in front of the wall is responsible to some extent. In any case, the low end is definitely for real as witnessed by the ability of these speakers to shake the walls and floor!

I have not yet run REW to generate room compensating filters beyond my manually generated ones. So far, I am very impressed with the performance of these speakers and can listen to them for hours as is. Sorry, I'm not going to get all flowery and unbelievable about soundstage, imaging, and transparency at this point, but I will say the bass performance of the Dayton Audio RSS315HO44 drivers is astonishing. My wife says she can always tell when I have played them loud while she was gone because all the pictures in the living room are hanging crooked!

Next Steps

I plan to do further tests and filter tuning including variations in crossover filter parameters, adding delays determined from driver measurements, and crossover designs with less steep attenuation as well as alternative use of shelf filters. The goal being improvements in phase linearity and better time alignment. This should enable achieving a cleaner impulse response, and thus yet better imaging and rendition of the original performances.

One additional design element: When it came time to run wires from the amplifiers to the speakers, the shortest and neatest route was through the floor and across in the basement. Given the thickness of good speaker cable, I was not happy with the idea of drilling large holes in the floor. The solution was to drill smaller holes through the baseboard quarter round and lining them with brass tubing sized and spaced for standard banana plugs. On the living room side, these give a neat appearance (**Photo 9**), and in the basement, the speaker wires are crimped and soldered into the brass tubes.

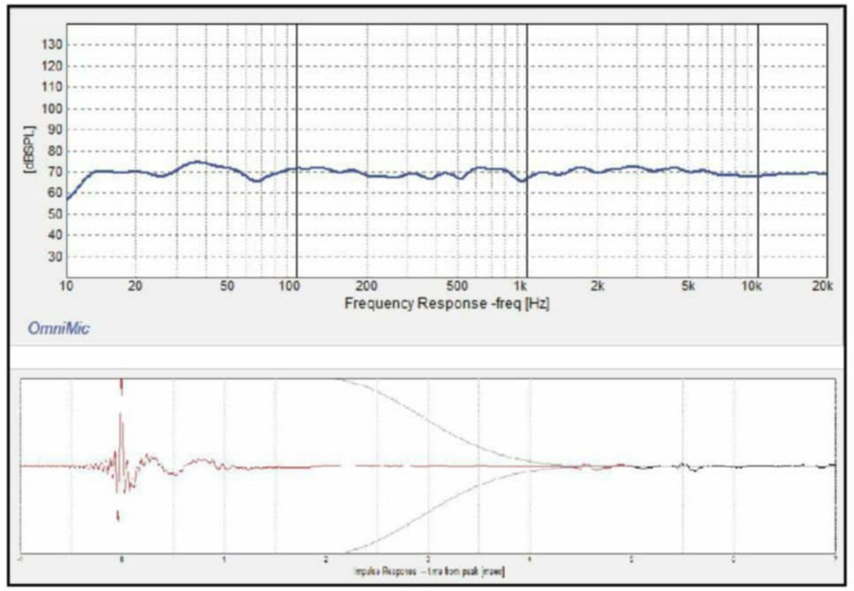


Figure 3: This graph shows the measured frequency at the typical listening position with 1/12th octave smoothing, and the impulse response measured 5′ in front of the speaker.

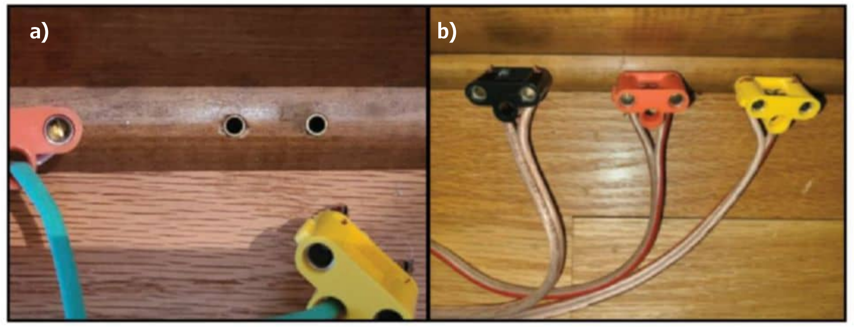


Photo 9: On the living room side, these give a neat appearance (a); and in the basement, the speaker wires are crimped and soldered into the brass tubes (b).

Recources

B. Cordell and R. Savas, "The Dh-220C MOSFET Power Amplifier," (Part 1): The Circuit, *audioXpress*, July 2021, https://audioxpress.com/article/you-can-diy-the-dh-220c-mosfet-power-amplifier-part-1-the-circuit

B. Cordell and R. Savas, "The Dh-220C MOSFET Power Amplifier," (Part 2): Construction and Testing Instructions, *audioXpress*, August 2021, https://audioxpress.com/article/you-can-diy-the-dh-220c-mosfet-power-amplifier-part-2-construction-and-testing-instructions

W. Maas, "A Visit to Purifi Audio: Where the Magic Happens, *audioXpress*, July 2020.

T. Perazella, "The Incredible Egg (Part 1): Sound Reproduction at Its Best," *audioXpress*, September 2018.

T. Perazella, "The Incredible Egg (Part 2): Assembling the Eggs," *audioXpress*, October 2018.

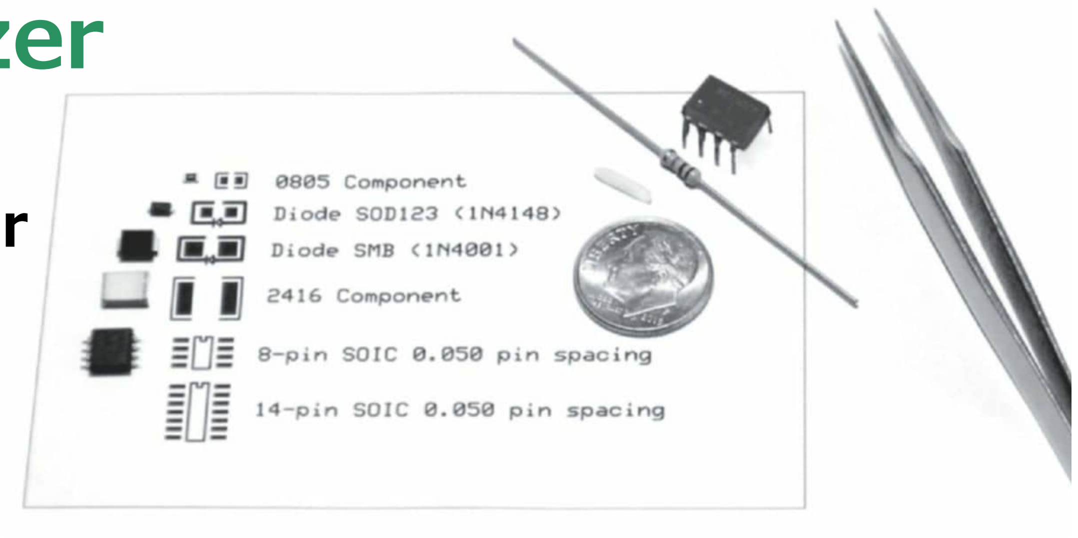
T. Perazella, "The Incredible Egg (Part 3): The Proof Is in the Listening," *audioXpress*, November 2018.

V. Dickason, *Loudspeaker Design Cookbook, 7th Edition*, 2006, available from KCK Media, cc-webshop.com

Building a Guitar-Controlled Synthesizer

Input Section & Pick Detector

By
Ethan Winer



The three previous articles described circuits that Ethan Winer developed for his current project that uses an electric guitar instead of a keyboard to control an analog synthesizer. This article continues with two pieces needed to process input from an electric guitar: the input section's preamplifier and noise gate, and the pick detector.

There are two pieces needed to process input from an electric guitar: the input section's preamplifier and noise gate, and the pick detector. The preamp amplifies the weak signal from an electric guitar's pickup and brings it up to line level for further processing. The noise gate detects when a note is currently playing and when it stops. Of course, the pick detector signals each time a new note is picked.

As with the previous synthesizer components in this series, the circuits in this article were developed using the LTspice program available for free from IC maker Analog Devices. One terrific feature of LTspice is its ability to accept input from a .wav file, and also output voltages as audio to a .wav file. Under Project Files is a link to a Zip file with the LTspice circuit file shown in **Figure 1**, plus the small .wav file this circuit expects as input.

Input Section

In LTspice, full scale (0 dBFS) in an input .wav file comes into the circuit at ±1V or 2V peak-to-peak. This is shown in the **Figure 2** screen-capture from Sound Forge, the audio editing software I use. Passive electric guitar pickups output somewhere around 250mV, depending on many factors. So the example .wav file this circuit requires was set to output that amount from voltage source V3. Note the wave file metacommand at the top left of the schematic. This is how you tell LTspice to use a file for input rather than a sine wave or other more common input test signals.

Passive guitar pickups sound best, and give the flattest response, when driving a high impedance input. Most guitar-based effects have an input impedance of at least 1MΩ (one million ohms), but even higher never hurts. So R1 in the input preamp is 3.9M, and op-amp U1 is an FET-type, which further ensures a high input impedance. LTspice doesn't have a potentiometer component, so resistor R3 had to be drawn that way using line art. In the real circuit R3 will be a user-accessible 100kΩ potentiometer that sets the overall system gain. In practice this could be either an internal trimpot, or an external control if the device will be used with more than one electric guitar. With the R2 through R4 values shown, the preamp has a gain range of about 5dB to 20dB.

After the preamp, the guitar signal passes through a very steep 50Hz four-pole high-pass filter to remove the "thumps" that can happen when a guitar player rests his hands on the strings. The lowest note with standard guitar tuning is the low E at 82.4Hz, so this filter's frequency is too low to affect that. The 220kΩ resistor R9 provides a small amount of positive feedback to help flatten the pass-band response.

The Thump filter then feeds a full-wave no-drop rectifier that outputs a positive-going version of the signal whether the input is positive or negative. Musical instrument waveforms are often asymmetrical, and guitar players can pluck a string either up or down, so this circuit must be able to respond to ether polarity of the input. A normal full-wave rectifier found in power supplies loses 0.6V across each rectifying diode, and that would be unacceptable here given the relatively small signal voltages. Placing the diodes within the feedback loop of op-amp U5 avoids that voltage drop. Op-amp U6 then applies 20dB additional gain via R14 to bring the signal up even higher for more accurate processing in the sections that follow. (For unity gain in the rectifier R14 would be 20kΩ instead of 200kΩ.)

Noise Gate Level Detector & Peak Follower

Next, the rectified guitar signal passes through two voltage comparators. Guitar notes tend to fade

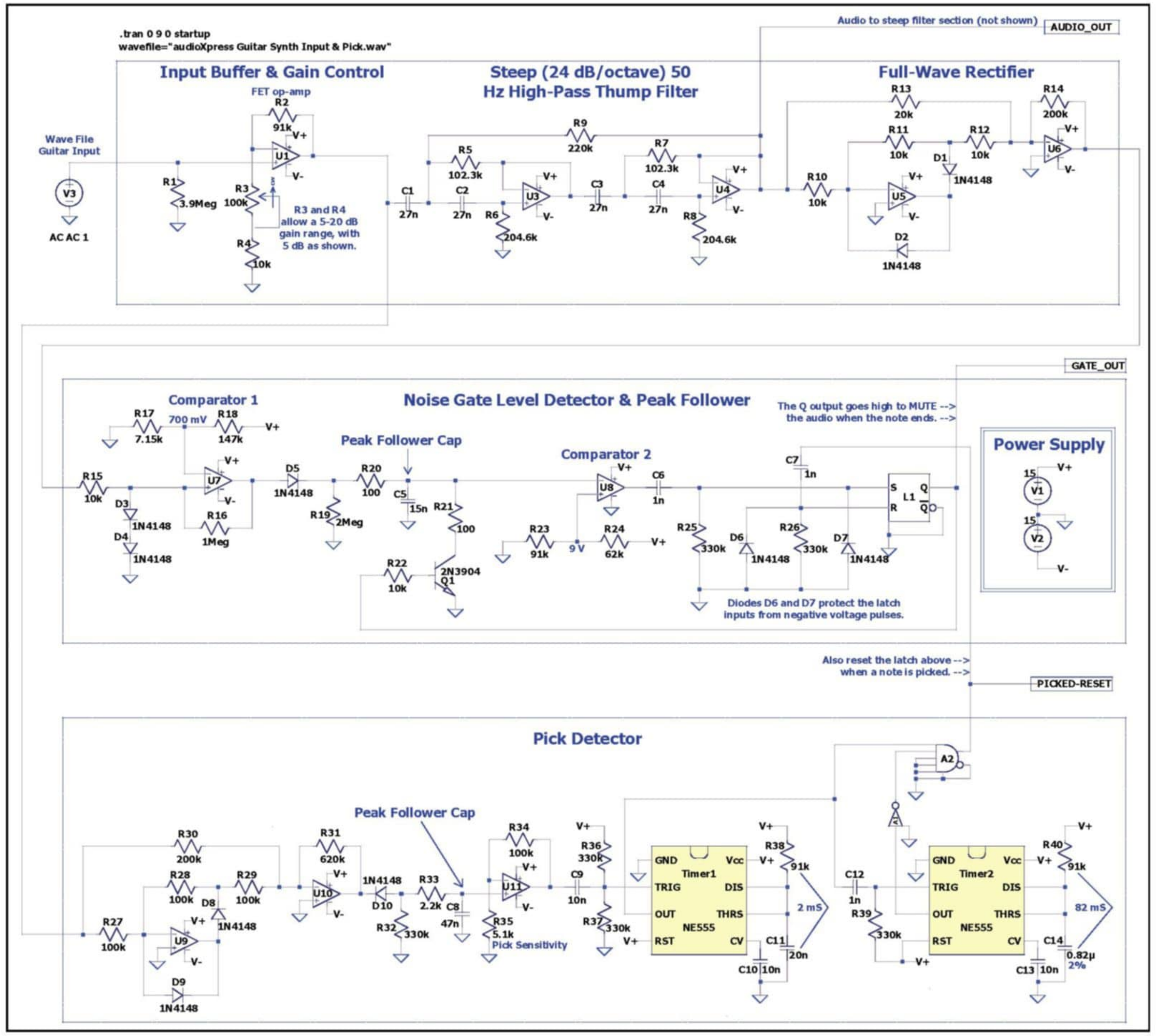


Figure 1: The full schematic for these circuits is divided into three sections (plus the power supply), with boxes around each section for clarity.

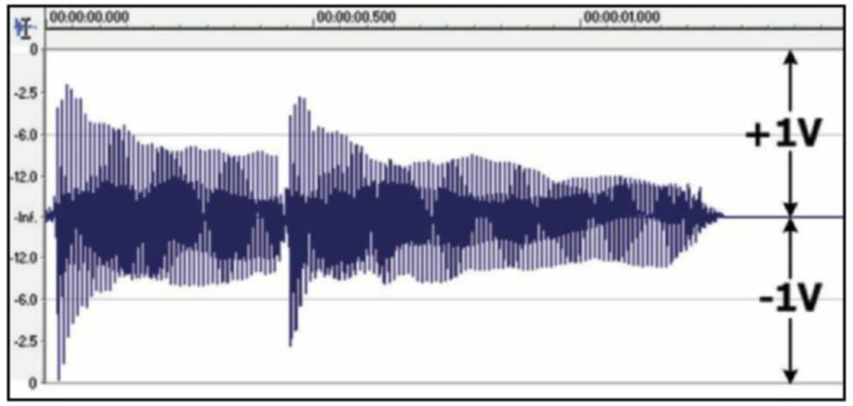


Figure 2: Wave files used for both input and output in LTspice treat 0 dBFS as 1V peak in each direction, or 2V peak-to-peak.

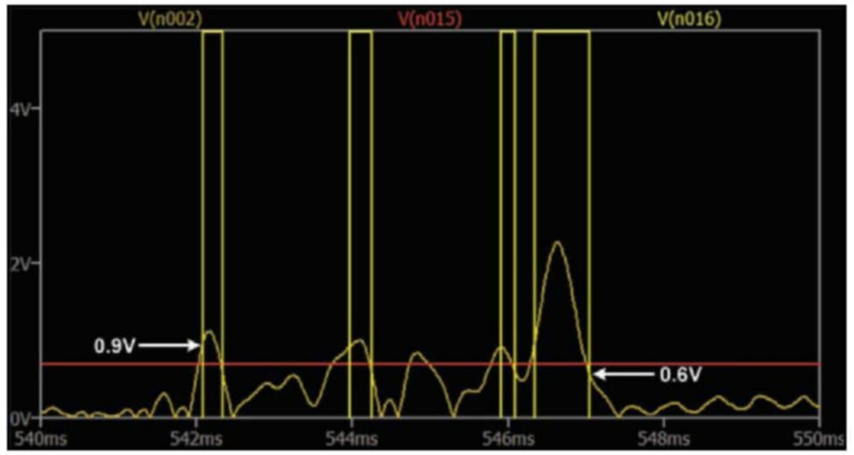


Figure 3: I drew flat tops on the yellow traces for clarity because the display is zoomed in to highlight only the bottom 5V, and the real tops extended way up off the screen. The bottoms of those traces also go off-screen, to -15V.

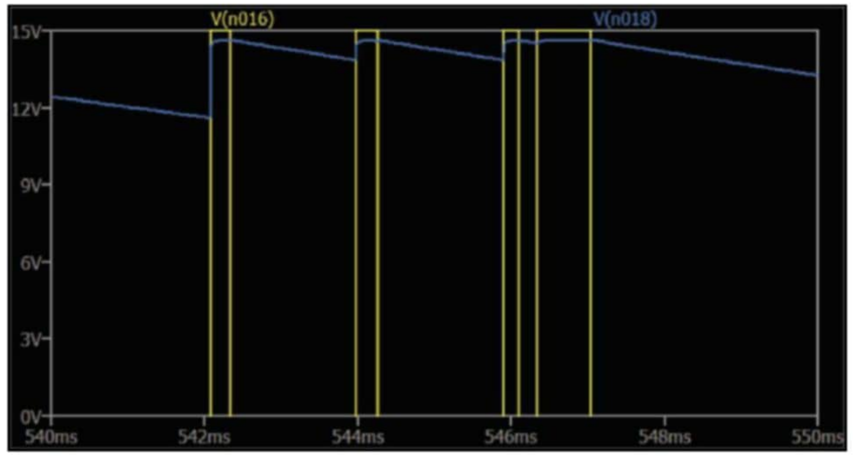


Figure 4: Every time Comparator 1's output changes to positive on the incoming guitar wave cycles, the Peak Follower capacitor, shown in blue, is recharged.

out quickly, so a pair of "clipping" diodes at the input to U7 limit the waveform peaks to around 1.2V, which keeps the voltages more uniform as they decay and thus easier to process. Since op-amp comparator U7 outputs a healthy 15V, the small drop in diode D5 is irrelevant.

My earlier article (*audioXpress*, June 2022) showing the LFO and ADSR modules explained the concept of hysteresis, whereby a controlled amount of positive feedback is applied to preserve a system's state. In that article I used as an example, "Don't turn off the furnace until you get slightly above the temperature I set." That LFO circuit uses a large amount of hysteresis around an op-amp comparator to generate its large square wave output. The comparator doesn't flip states until the input voltage is almost as high as the power supply! In the Noise Gate shown here, op-amp comparator U7 uses only a small amount of hysteresis via R15 and R16, to minimize the comparator constantly flipping states within each guitar wave cycle.

A peak follower circuit usually charges a capacitor with a varying voltage through a diode, and that voltage is buffered by an op-amp to isolate the capacitor from being drained by the circuits that follow. In this case the input is the fixed-level output from comparator U7, and the next circuit element is an op-amp comparator that serves as a buffer. So the capacitor alone can be considered the peak follower.

Where Comparator 1 in U7 assesses the individual cycles in the incoming guitar wave, Comparator 2 in U8 instead looks at the larger picture to identify each note's On and Off times. **Figure 3** probes several places in the circuit at once to better show how they all interact. The brown trace is the guitar wave at the output of U6 after rectification, so it extends in the positive direction only. The red trace is the 0.7V loudness reference for Comparator 1, and the yellow trace is the output of Comparator 1 U7. Even though the reference level for Comparator 1 is set to 0.7V, its output doesn't flip positive until the input level reaches 0.9V due to the hysteresis set by resistors R15 and R16. Then when the guitar wave drops back down, the comparator's output doesn't go negative until the voltage drops to below 0.6V.

Figure 4 shows the output of Comparator 1 in yellow, but this time along with the voltage at Peak Follower capacitor C5 in blue. As long as the comparator continues to switch positive on the incoming guitar wave peaks, C5 keeps getting pumped up toward the 15V power supply. That C5 voltage then goes to Comparator 2, whose output is shown as red in **Figure 5**. Here, the virtual oscilloscope is zoomed way out to see entire note durations rather than individual wave cycles. As the guitar's current note fades out and C5 stops being charged, Comparator 2 waits for C5 in blue to fall below 9V. At that point the note is considered to have ended and Comparator 2 flips positive.

The times and the comparator voltage levels in these circuits are somewhat critical. If the peak follower's decay time is fast (if R19 is smaller), Comparator 2 can "chatter" (quickly flip back and forth repeatedly) even though there is some hysteresis in Comparator 1. But if the decay time is too slow, noise residue at the end of a fading note continues to get through until the cap drains fully. To avoid this, when Comparator 2 decides the note has ended, it sets latch L1, which dumps the peak follower capacitor, forcing a low input level to Comparator 2 to avoid repeated switching. The latch is then reset the next time a note is picked. (The sidebar briefly explains digital latches.) Note that Comparator 1 needs to run on ±15V rather than just +15V to keep the up and down hysteresis thresholds more symmetrical.

Pick Detector

Like the Noise Gate that needs to recognize both positive and negative voltage peaks, the pick detector also needs a full-wave rectifier. I probably could have used the same rectifier circuit that's in the gate section, but this one needs to output a negative voltage and it requires less gain. It also seems to work better taking its input from before the 50Hz high-pass "thump" filter. Electronic components are cheap, so I just added a new separate circuit. This rectifier drives a second peak follower capacitor, C8, whose buffer op-amp adds yet more gain. That triggers a 555 timer IC to create the 2mS pulse signaling that a note was picked on the guitar.

A 555 timer chip begins its cycle of outputting a positive voltage when the Trigger input transits from above to below one-third the supply voltage. So with a 15V power supply the trigger must extend down to below 5V. This is why more gain is needed in op-amp U11. But the trigger must be a brief pulse that ends before the timer's period completes, otherwise the timer's output will remain high. That is, a negative-going trigger starts the timing cycle, but if the trigger remains low the timer continues its positive output past its set time. The solution is the differentiator created with C9, R36, and R37.

Most differentiators convert a voltage step into a brief pulse using a series capacitor and one resistor tied either to ground or the power supply, depending on the required pulse direction. **Figure 6** shows the

Digital Latches

A latch is a clever and useful digital logic element. Like the Data flip-flops used in the Sample & Hold and Period-To-Voltage Converter circuits shown previously, a latch can serve as one byte of memory. But a latch is more simple because it doesn't need a Clock input to tell it when to store the data. You simply apply a logic 1 to its Set input "S" to store one bit, or to its Reset input "R" to clear its output to 0. And like the D-flop, it has both Q (normal) and Q-Bar (inverted) outputs. In the Noise Gate circuit, a latch is used to stabilize the gate's output, so it won't go on and off between the incoming guitar wave cycles. And like the 555 timers, both inputs are driven with a brief differentiator pulse to ensure that sustained input voltages are ignored.

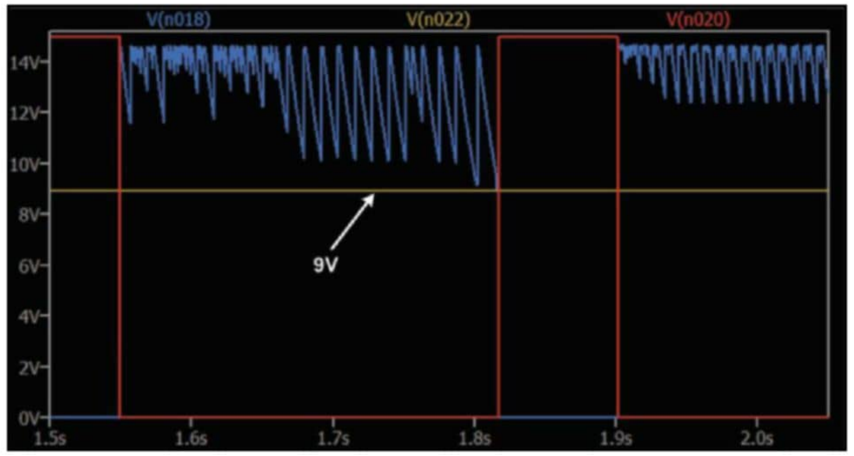


Figure 5: Comparator 2 monitors the falling output from Peak Follower capacitor C5 shown in blue, finally flipping positive when the guitar note has faded out sufficiently.

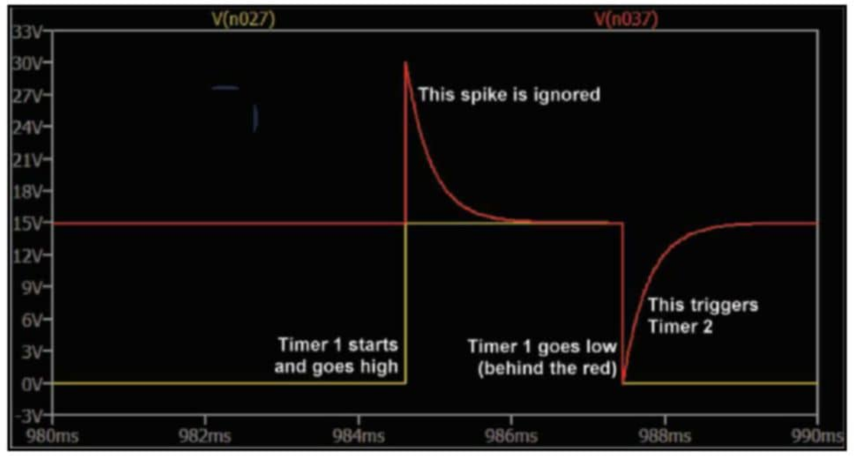


Figure 6: A differentiator changes the brown input voltage step into the shorter output pulse shown in red. The resistor and capacitor values determine the pulse durations.

About the Author

Ethan Winer has been an audio engineer and professional musician for more than 45 years. His Cello Rondo music video has received nearly 2 million views on YouTube and other websites, and his book *The Audio Expert* published by Focal Press, now in its second edition, is available at amazon.com and Ethan's own website ethanwiner.com. Ethan is also a principal at RealTraps, a US manufacturer of high-quality acoustic treatment.

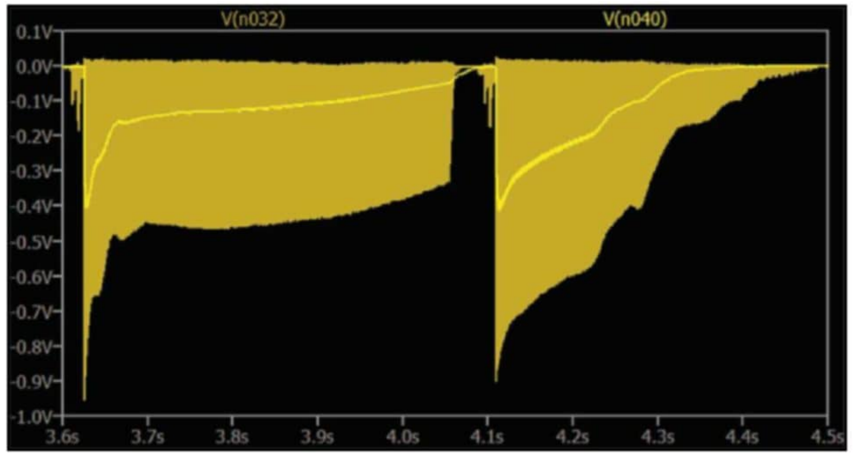


Figure 7: When scaled to show two entire guitar notes, you can't see the individual wave cycles at the output of rectifier U10 in brown. But the filtered output in yellow at peak follower capacitor C8 is starting to take shape.

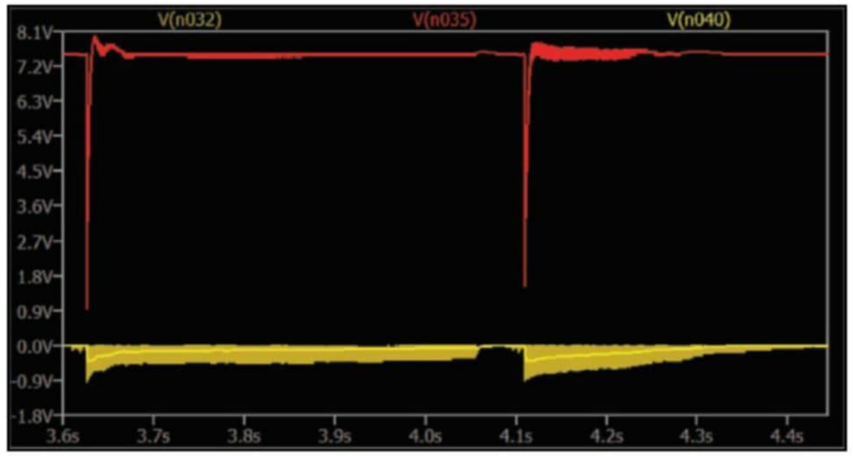


Figure 8: Now we're getting somewhere! Here, the rectifier's output is again shown in brown, this time along with the Trigger input to Timer 1 in red. After being amplified by U11, and going through the differentiator formed by C9, R36, and R37, we now have a distinct signal pulse capable of triggering a 555 timer.

Project Files

To download the links for LTspice circuit file and input Wave file, visit: http://audioxpress.com/page/audioXpress-Supplementary-Material.html

Recources

LTspice, www.analog.com/en/design-center/design-tools-and-calculators/ltspice-simulator.html

LTspice online support forum, https://groups.io/g/LTspice

left and right sides of capacitor C12 that triggers Timer 2 when Timer 1's output goes negative after its 2mS period ends. Note that Timer 1's Trigger input is biased to rest at half the supply voltage. This way the triggering signal can be as small as 2.5V. Here's another clever little hack: To simplify calculating 555 times I use 91kΩ for the pull-up resistors (R38 and R40); then the timing capacitor's value (C11 and C14) = mS/10 nF. For low-power circuits you can use 910kΩ, then mS= nF. Using "91" also works for microfarads (μF) as you can see in Timer2.

Figure 7 shows the output of rectifier U10 in brown, zoomed out to see two entire guitar notes. The voltage at Peak Follower capacitor C8 shown in yellow is 0.6V less due to the voltage drop across diode D10. **Figure 8** adds to those displays the voltage at Timer 1's Trigger input in red, after amplification and differentiation. This is now suitable for triggering Timer 1 to create the 2mS Picked pulse needed to start a note playing elsewhere in the synthesizer. But wait, we're not quite done.

The initial few cycles of a picked guitar note are much larger than subsequent cycles as the note fades out. So it's possible for Timer 1 to be triggered quickly two or three times in a row on those initial large wave cycles. To prevent this, Timer 2 is set for 82mS, and along with the AND gate serves to lock out additional Picked output pulses.

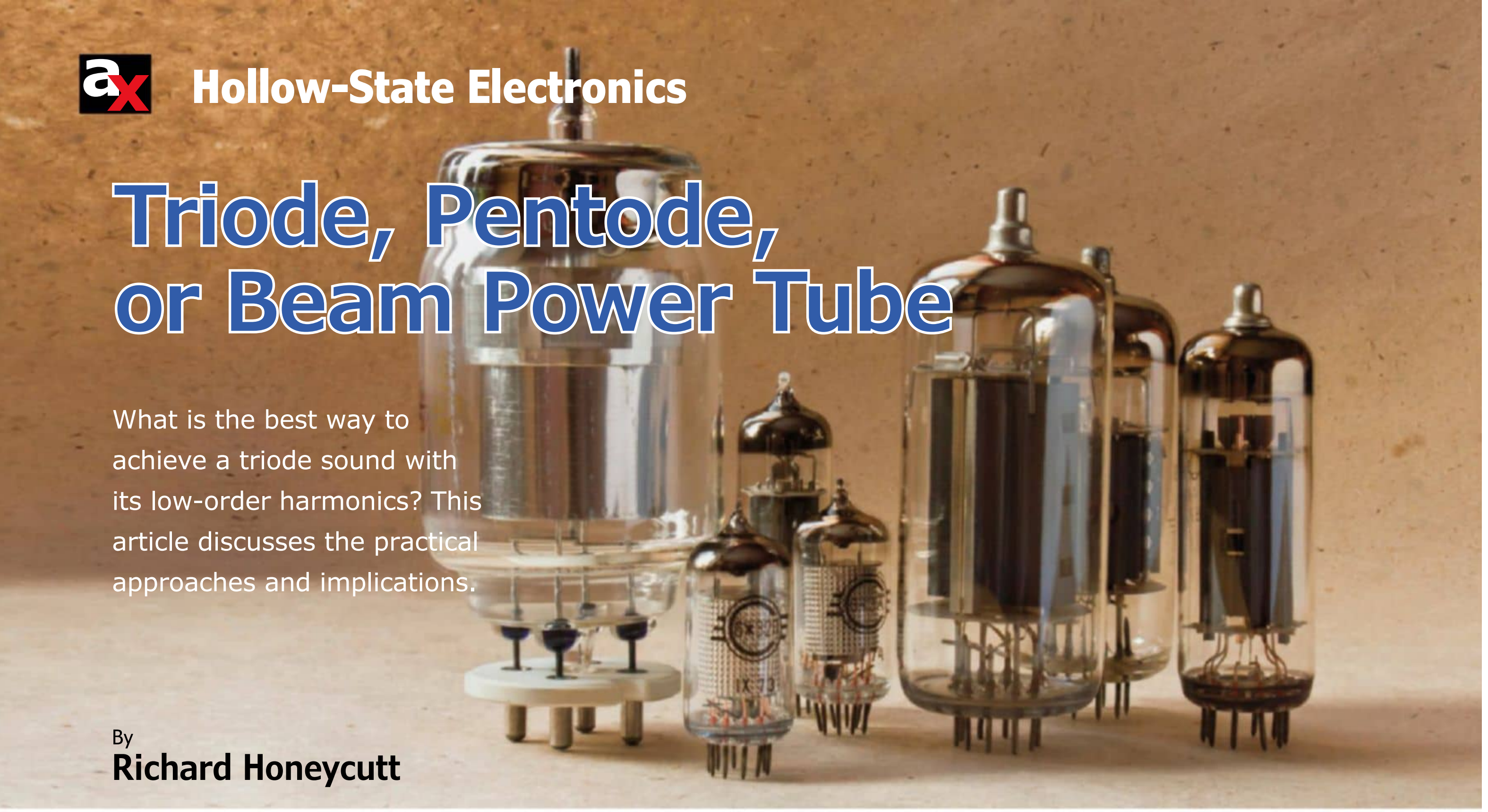
The logic is pretty simple: At rest, both Timer 1 and Timer 2 output 0 volts. But Timer 2 goes through an inverter, so that input to the AND gate is active, or "1." Then when a note is picked and Timer 1 goes high, the AND gate's output also goes high. But 2mS later Timer 1 completes, which triggers Timer 2 to start, so its inverted output goes negative disabling the AND gate for 82mS. This delay is long enough to avoid double-triggering, but still allows picking 16th notes at a tempo of 183 beats per minute (BPM), which is some very fast picking!

Finally, as explained in the previous articles, unused inputs and outputs of LTspice logic elements should be grounded. This avoids the program wasting time calculating connections that aren't used. But once the LTspice simulation works as expected and you build the circuit for real, the inverting Q-Bar output of Latch L1, and the "NOT" output of AND gate A2, must be left floating, not grounded. This is also why three of the AND gate inputs are grounded. In this case they are disabled, not set to Logic 0.

Editor's Note: All audioXpress *articles from 2001 to present can be found on the aX Cache, a USB drive available from www.cc-webshop.com.*

Triode, Pentode, or Beam Power Tube

What is the best way to achieve a triode sound with its low-order harmonics? This article discusses the practical approaches and implications.



By
Richard Honeycutt

DIY hollow-state power amp designer/builders have a number of choices for output tubes, as we discussed in the May 2013 Hollow-State Electronics article. Rather than repeat these discussions mentioning individual tubes and tube families, this article will examine the more generic issue: If we want a "triode sound" with its low-order harmonics, does it matter whether we achieve it by using a "triode-connected" pentode or beam power tube rather than a true triode? And, what are the performance downsides of using a triode connected beam power tube or pentode rather than sticking with the pentode connection?

Figure 1 shows a very simple Class A1 push-pull output stage using 300B tubes. The parts values are chosen according to the recommendations for Class A1 operation on the original Western Electric datasheet shown in **Table 1**.

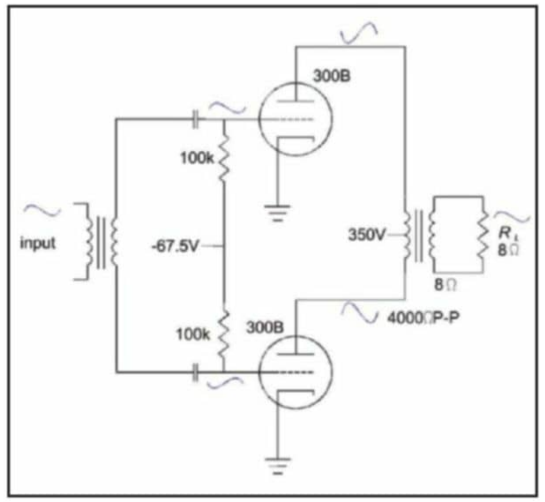


Figure 1: This simple push-pull triode power amp uses the classic 300B tubes.

PUSH-PULL AMPLIFIER — CLASS A_1
Unless otherwise specified, values are for 2 tubes

Filament Voltage, A-C	5.0	5.0	volts
Plate Voltage	300	350	volts
Grid Voltage*	−61	−67.5	volts
Peak A-F Grid-to-Grid Voltage	122	135	volts
Zero Signal Plate Current	100	170	milliamperes
Maximum Signal Plate Current	150	200	milliamperes
Effective Load Resistance (Plate-to-Plate)	4000	4000	ohms
Maximum Signal Power Output	10	20	watts
Total Harmonic Distortion	4.5	2	per cent

Table 1: The parts values are chosen according to the recommendations for Class A1 operation on the original Western Electric datasheet.

LTSpice modeling of the circuit shows that an input of 175.6 V_{PEAK} is required to produce an output of 20W, not 135V as indicated on the datasheet. At this output level, fast Fourier transform (FFT) analysis (**Photo 1**) shows 1.1% third harmonic, and 1.7% fifth harmonic, for a THD of 2.3%. Since this is Class A1 push-pull, the maximum output power is exactly twice as much as that of a single-ended 300B triode power amplifier. If it were Class AB1 or AB2, this pair of triodes could put out at least 30W.

Beam Power Tube in a Push-Pull Class A1

Figure 2 shows an LTSpice model of a tetrode-connected 6550 pair in a Class A1 push-pull stage. The parts values correspond to those shown for this type of circuit in the RCA datasheet (**Table 2**). However, there is quite a bit of confusion online concerning the 6550 and its European counterpart, the KT88 (**Photo 2**).

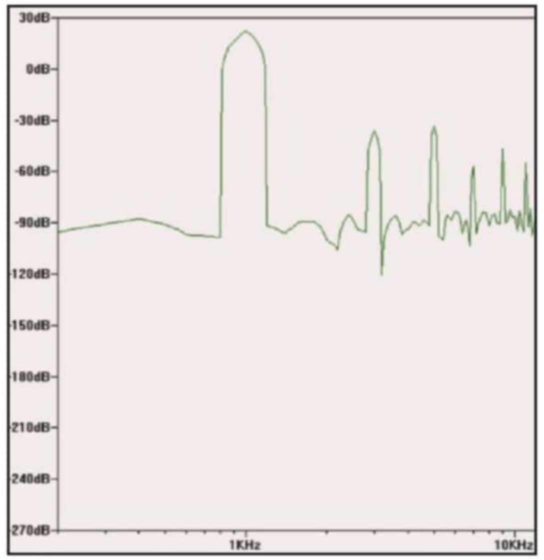


Photo 1: The FFT analysis of the output from the LTSpice model of the Class A1 300B push-pull circuit shows THD of 2.3%.

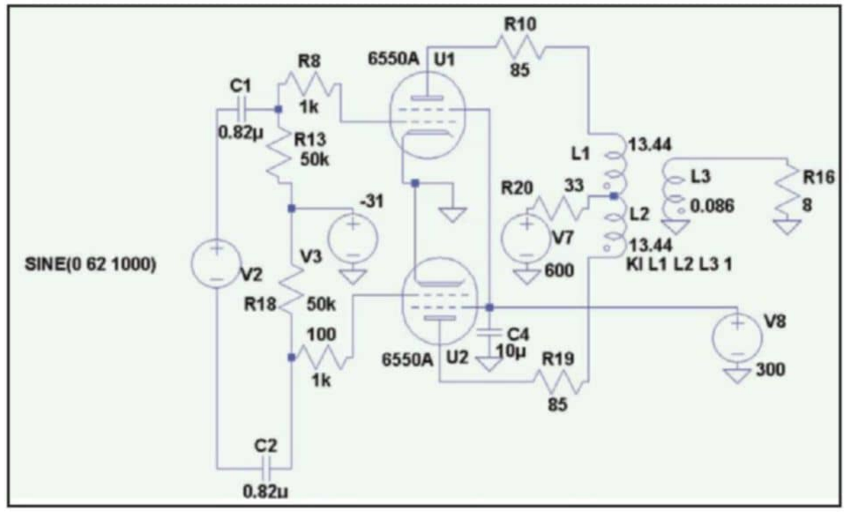


Figure 2: This is the LTSpice model of a Class-A1 push-pull stage using 6550 tubes in the tetrode connection.

The first audio power tubes manufactured were power triodes. Some power tetrodes have been used over the years, but early tetrodes were subject to self-oscillation due to a negative-resistance "kink" in their plate characteristic curves. In 1935, J. Owen Harries discovered that proper spacing of the plate with respect to the screen-grid-to-cathode distance would maximize the efficiency of a power pentode and reduce secondary emission. Since the "tetrode kink" was produced by secondary emission, the "Harries Valve" eliminated the kink. By this time, the power pentode had been invented and patented by Philips. Having a higher voltage gain than power triodes, they quickly began to be adopted.

In 1936, RCA developed the 6L6 beam power tube, in which Harries' innovation was accompanied by replacing the suppressor grid with "beam-forming plates." This design worked very well, and

Typical Operation and Characteristics:
Values are for 2 tubes

	Fixed Bias		Cathode Bias	
Plate Supply Voltage	400	600	400	volts
Grid-No.2 Supply Voltage	275	300	300	volts
Grid-No.1 Voltage	-23	-31	–	volts
Cathode Resistor	–	–	140	ohms
Peak AF Grid-No.1-to-Grid-No.1 Voltage	46	62	53	volts
Zero-Signal Plate Current	180	115	166	ma
Max.-Signal Plate Current	270	273	190	ma
Zero-Signal Grid-No.2 Current	9	4	7.5	ma
Max.-Signal Grid-No.2 Current	44	41	39	ma
Effective Load Resistance (Plate to plate)	3500	5000	4500	ohms
Total Harmonic Distortion	3	2.5	4	%
Max.-Signal Power Output	55	100	41	watts

Maximum Circuit Values:

Grid-No.1-Circuit Resistance:		
For fixed-bias operation	0.05 max.	megohm
For cathode-bias operation	0.25 max.	megohm

Table 2: The parts values correspond to those shown for this type of circuit in the RCA datasheet.

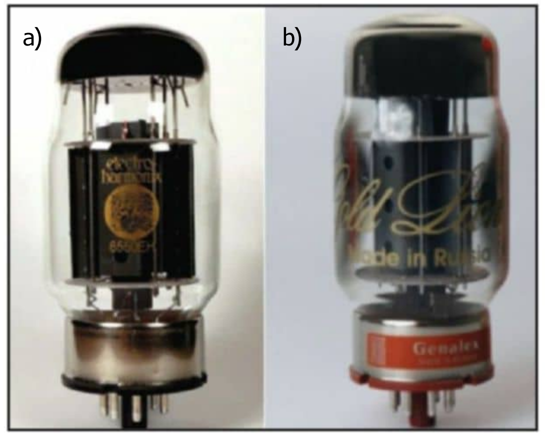


Photo 2: The 6550 beam power tube (a) and the "kinkless tetrode" KT88 (b) are functionally identical.

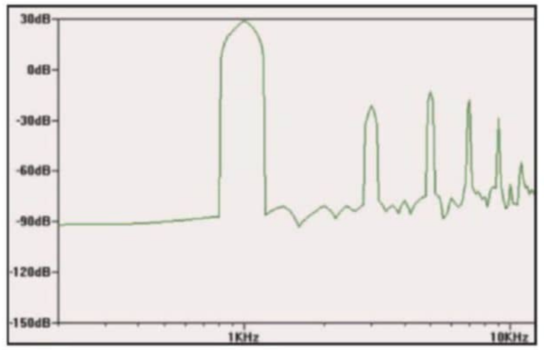


Photo 3: The FFT analysis of the output from the LTSpice model of the Class A1 6550 push-pull circuit (tetrode connection) shows THD of 1.16%.

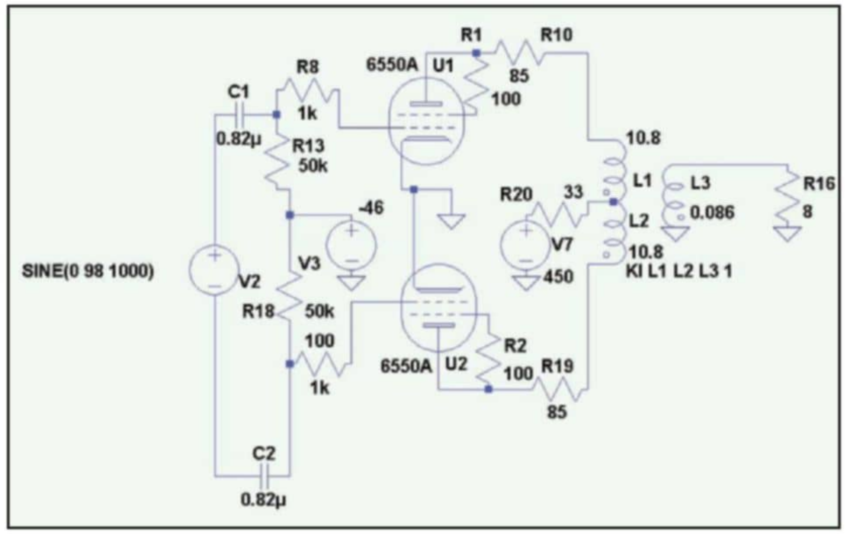


Figure 3: The 6550 connected as a Class A1 push-pull pair of triodes can output 27W with a THD just under 1.3%.

PUSH-PULL AF POWER AMPLIFIER — Class A1

Triode Connection[d]

Maximum Ratings, *Design-Center Values:*		
PLATE VOLTAGE	495 max.	volts
GRID-No.2 (SCREEN-GRID) VOLTAGE	440 max.	volts
GRID-No.1 (CONTROL-GRID) VOLTAGE:		
Negative-bias value	330 max.	volts
Positive-bias value	0 max.	volts
CATHODE CURRENT	192.5 max.	ma
GRID-No.2 INPUT	6.6 max.	watts
PLATE DISSIPATION	44 max.	watts
BULB TEMPERATURE (At hottest point on bulb surface)	250 max.	°C
Typical Operation and Characteristics:		
Values are for 2 tubes		
Plate Voltage	450	volts
Grid No.1 Voltage	-46	volts
Peak AF Grid-No.1-to-Grid-No.1-Voltage	92	volts
Zero-Signal Plate Current	150	ma
Max.-Signal Plate Current	220	ma
Effective Load Resistance (Plate to plate)	4000	ohms
Total Harmonic Distortion	2.5	%
Max.-Signal Power Output	28	watts
Maximum Circuit Values:		
Grid-No.1-Circuit Resistance:		
For fixed-bias operation	0.05 max.	megohm
For cathode-bias operation	0.25 max.	megohm

Table 3: The LTSpice model of a triode-connected 6550 push-pull pair with the circuit values shown in the RCA datasheet.

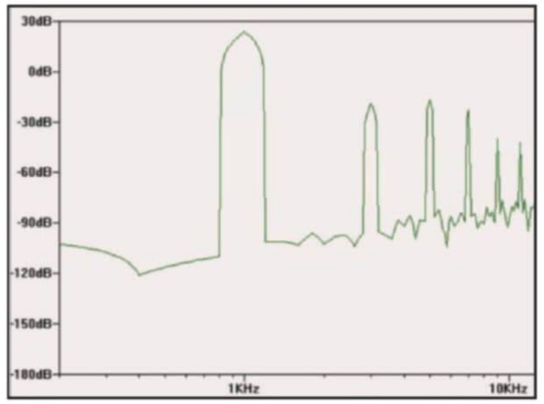
since the 6L6 is a "beam power tetrode," it evaded the Philips patent on power pentodes. The KT66 ("kinkless tetrode 66") is the European version of the 6L6GC. The 6550 is a higher-voltage, higher-power tube similar to a 6L6GC. It is rated at 600V rather than the 500V of the 6L6GC, and 35W plate dissipation rather than the 30W of the 6L6GC.

Its European counterpart is the KT88, which is almost identical, but is rated at 40W plate dissipation. The comparison of ratings between the 6550 and the KT88 depends upon the manufacturer's datasheet you use. Some show slightly higher maximum plate voltage, plate current, or plate dissipation for one or the other of these tubes.

Electro-Harmonix publishes a datasheet identified as being for the "6550EH, KT88EH, KT88SV, KT88 Genalex" tubes. This datasheet shows no difference in specs among these tube types. The Svetlana KT88 datasheet says that its KT88 is an exact replacement for the 6550, 6550A, KT88, KT90, KT99, and KT100. At least one online commenter has pointed out that the KT88 is a kinkless tetrode, while the 6550 is a beam power tube. However the internal construction may or may not vary, so these two power tubes can be considered functionally identical.

Photo 3 shows the FFT of the output wave of a 6550 pair connected as a Class A1 push-pull stage, using the circuit shown in Figure 2. The RCA datasheet shows an output power of 100W when the B+ voltage is 600V and the peak grid-to-grid input voltage is 62V. The THD shown under these

Photo 4: The FFT of the output wave from the circuit shown in Figure 3 shows significant odd-order harmonics.



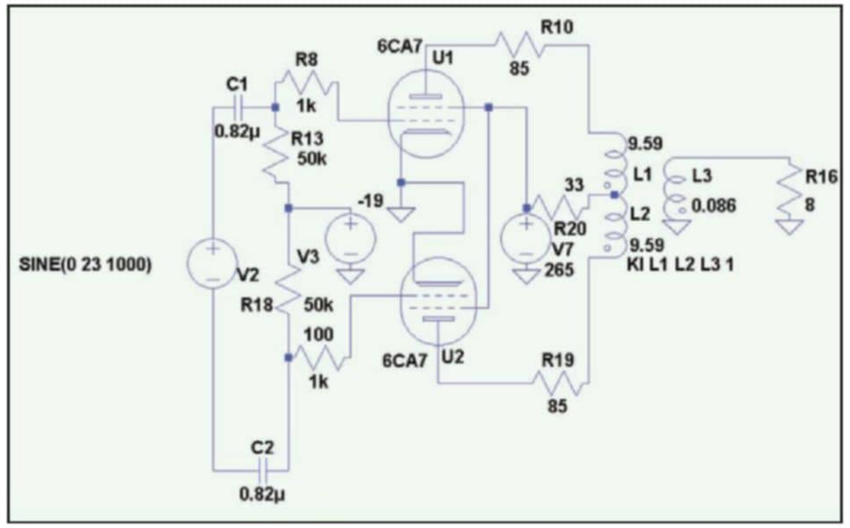
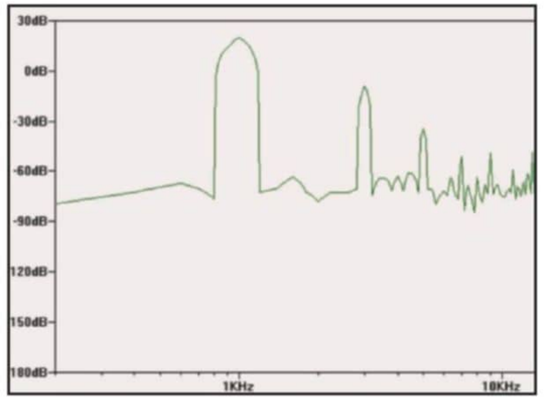


Figure 4: Two 6CA7s in Class A1 push-pull can provide 12W output at a THD of 3.6% using this circuit.

Photo 5: The 6CA7 in Class A1 push-pull produces almost no even-order harmonics.



A Pentode or beam power tube can be connected to perform as a triode. This can be accomplished by connecting the screen grid to the plate, either directly or through a small (~100Ω) resistor. Beam power tubes are often thought of as a type of pentode, but they were introduced as tetrodes, in order to avoid infringing on Phillips' patent on "Harries power pentodes." Since the beam forming plates accomplish the same function as suppressor grids, either designation is correct.

conditions is 2.5%. The LTSpice model shows an output power of about 100W with a THD of 1.16%: 0.88% H3, 0.66% H5, 0.4% H7, and 0.13% H9. Odd-order harmonics through the ninth can be clearly identified in the FFT.

Triode-Connected Beam Power Tube in a Push-Pull Class A1

Figure 3 shows an LTSpice model of a triode-connected 6550 push-pull pair with the circuit values shown in the RCA datasheet (**Table 3**). The LTSpice model required an input of $98V_{PEAK}$ to achieve an output of 27W. With this input voltage, third-harmonic distortion was 1.1%, fifth and seventh were each 0.5%, ninth was 0.32%, and THD was 1.3% in Class A1 push-pull (**Photo 4**).

Power Pentode in Push-Pull Class A1

Power pentodes behave similarly to beam power tubes. The beam forming plates of a 6550 perform equivalently to the suppressor grid of a 6CA7/EL34. However, I thought it would be interesting to see whether the similarity in performance extends to the triode-connected configuration of a power pentode. I could not find a datasheet with "typical operation and characteristics" for a power pentode. The Tung-Sol datasheet does give this information for "Class A – One Tube" and "Class B – Two Tubes." I created a Class A1 push-pull design using the 6CA7 based upon this data.

When modeling a power pentode for LTSpice, one would expect to use the "pentode" device outline. However, almost all available power pentode models have the suppressor grid connected internally to the cathode, and thus use the "tetrode" outline. **Figure 4** shows two 6CA7's connected in Class A1 push-pull. The circuit produces 12W with a THD of 3.6% composed almost entirely of odd harmonics, of which the third predominates (3.6%, with fifth harmonic at 0.19%), as you can see from **Photo 5**.

Using the 6CA7s in a triode connection drops maximum power to about 7W (**Figure 5**), with THD of 3.8% (3% third harmonic, 2.1% fifth harmonic, 0.9% seventh harmonic). Even-order harmonics are almost absent. **Photo 6** shows the FFT of the output wave.

The Bottom Line

Using the tetrode or pentode connection provides a higher maximum power output: 100W vs. 27W for the 6550 and 12W vs. 7W for the 6CA7. The pentode/tetrode's higher voltage gain reduces the output voltage that must be provided by the drive or phase splitter stages, potentially reducing distortion in those stages. The 6550's higher maximum plate dissipation (35W vs. 25W for the 6CA7 and cathode current (175mS vs. 150mA for the 6CA7) allows higher maximum output powers. Surprisingly, the RCA datasheet only provides "typical operation and characteristics" for the tube in Class A1—not Cass AB1 or AB2—push-pull circuits. The Tung-Sol datasheet does provide operating characteristics for the Class AB1 push-pull circuit using 6550s, but the maximum output power is still 100W, and the only difference between the Tung-Sol Class AB1 circuit and the RCA Class A1 circuit is that the former uses a control-grid bias of -32.5V, whereas the latter uses -31V.

Four points should be noted concerning harmonic distortion. First, the instantaneous signal level will control THD. Second, probably the most reliable test of the audibility of THD was performed by the Southwestern Michigan Woofer and Tweeter Marching Society (SMWTMS) [1]. In the complete series of A/B/X tests, THD was found to be audible at 4% using big band jazz music, 2% on flute music, and 0.4% on a sine wave. The spectrum of the harmonic distortion was not specified. Third, the distortion signature of a given circuit configuration depends less upon signal level than does THD; thus it probably constitutes a better metric for comparison. And fourth, THD can be reduced by using negative feedback or an ultra linear connection.

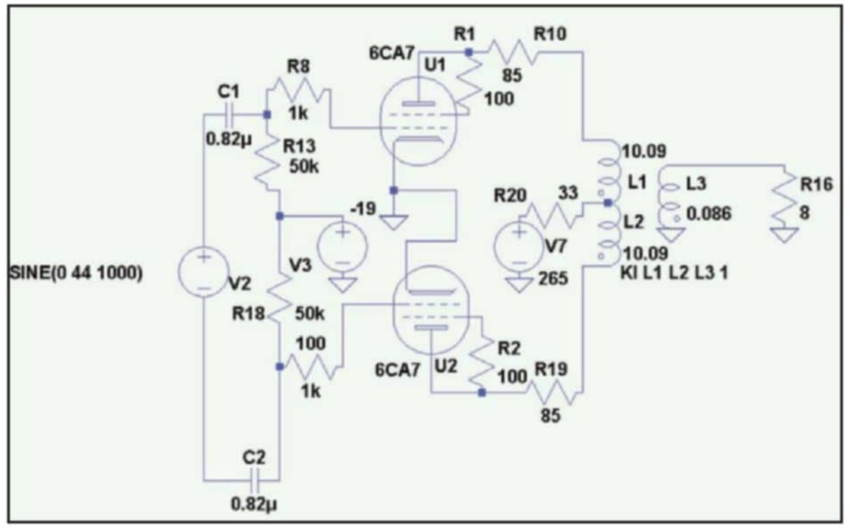
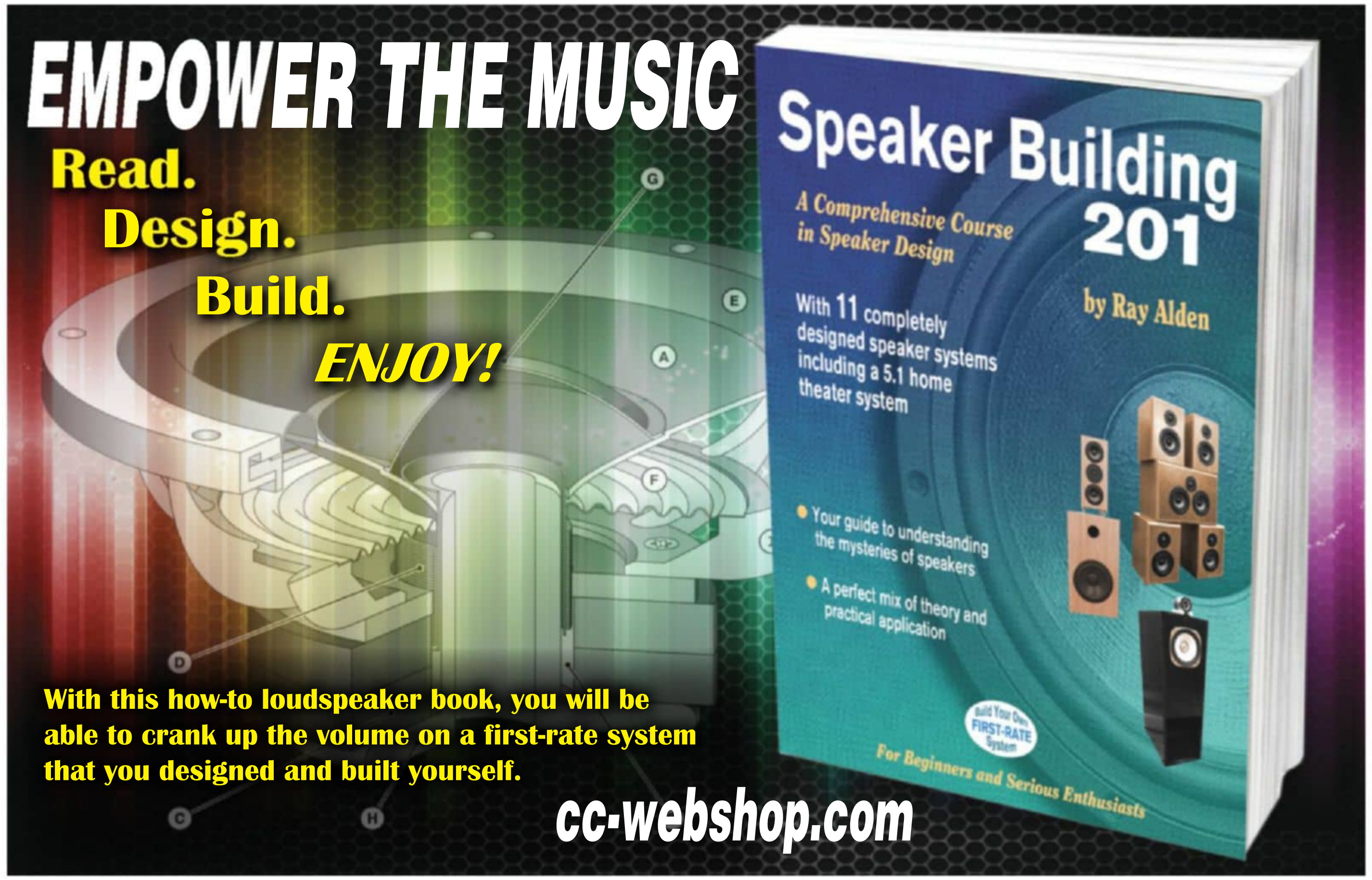


Figure 5: 6CA7s are connected as triodes in Class A1 push-pull can output 7.5W.



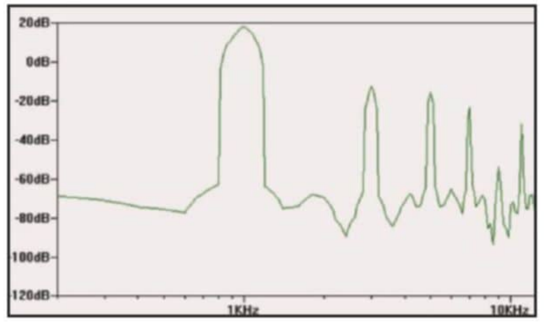


Photo 6: The triode-connected 6CA7 Class A1 push-pull pair has a THD of 1% at maximum output.

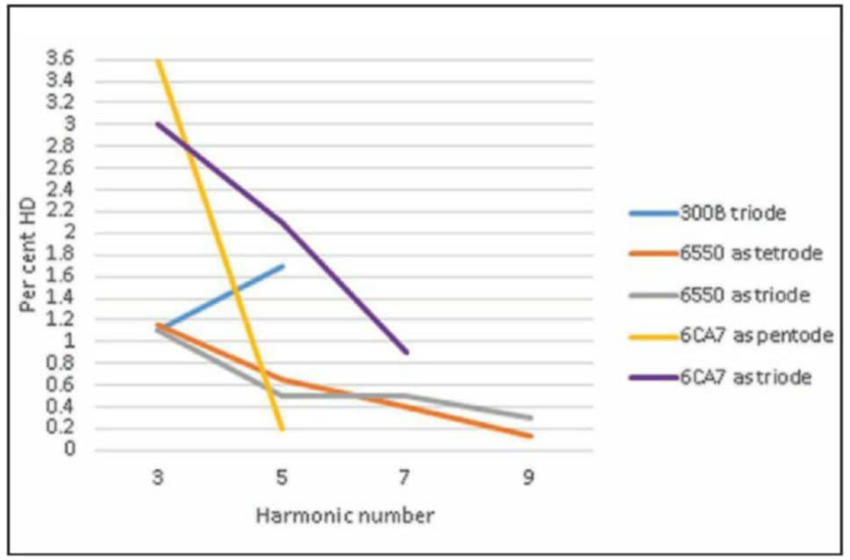


Figure 6: The distortion signatures of the different circuits vary widely.

About the Author

Dr. Richard Honeycutt bought a *Popular Electronics* magazine from the rack at a bus station in 1960. It contained an article on building a transistorized audio amplifier, which captured his interest. He followed up by subscribing to several electronics magazines. To avoid the cost of parts, he built up a "junk box" by disassembling trade-in TVs from his uncle's furniture store. His dad bought him the Van Valkenburg, Neville, and Nooger Basic Electronics book set, so he learned about tube circuits as well as transistors. He started repairing electronic devices about 1965, earning his First-Class Commercial FCC license in 1969. He has worked part-time in broadcast radio engineering and audio electronics repair since 1968, in addition to full-time work in acoustical and audio system design, plus a 20-year stint teaching electronics at the college level.

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[1] D. Clark, "Ten years of A/B/X Testing," Audio Engineering Society (AES) preprint 3167, October 1991.

Resources

"6550EH, KT88EH, KT88SV, KT88 Genalex," datasheet, Electro-Harmonix, https://shop.ehx.com/catalog/tubes-specs/KT88EH.pdf

"Svetlana KT88 Beam Power Tetrode" datasheet, Dr. Tube, www.drtube.com/datasheets/kt88-sed2001.pdf

Figure 6 shows the distortion signatures of the various circuits I simulated. As expected for push-pull circuits, even-order harmonic distortion is quite low. The 300B output wave shows high third-harmonic content, but—surprisingly— higher fifth harmonic, with negligible higher-order harmonics. The 6550 in the triode and tetrode configurations have quite similar distortion performance; both essentially match the third-harmonic levels of the 300B, but neither shows the increase in fifth harmonic compared to third, and both have significant harmonic components out to the ninth. It being generally accepted that high-order harmonics are more annoying than low-order ones, we would expect the 300B to sound better than the 6550 in either the triode or pentode configuration.

During the 1940s and 1950s, the soundtracks of military training films were often made using peak clippers (not true compressors), not only to prevent overdriving the amplifiers, but because a bit of added harmonic distortion was thought to help compensate for the poor high-frequency response of the projectors and speakers, thereby improving the speech intelligibility. The 300B's higher fifth-harmonic distortion could perhaps act in somewhat the same way, partially accounting for that tube's popularity, especially among older audiophiles whose hearing may lack high frequencies.

The 6CA7 has much higher third-harmonic distortion than the 300B or 6550, but the distortion drops rapidly at the fifth and seventh harmonics, becoming very low (triode configuration) or negligible (pentode configuration) above the seventh harmonic.

Based upon these simulations, there appears to be no general answer to the question of how much difference one could notice between using the pentode (or tetrode) connection for a beam power tube or power pentode versus the triode connection. The extent of the difference varies among different tubes, and its potential audibility could only be determine by careful listening tests.

It should be pointed out that the use of negative feedback would decrease the harmonic distortion in any of these circuits, although if the amplifiers were driven significantly harder (into hard clipping), very annoying harmonics well above the ninth could very well become problematic.

Pentodes or beam power tubes used in the ultra-linear circuit have sometimes been likened to power pentodes connected as triodes, although the similarity is not absolute. It would be interesting to compare the distortion signatures of this circuit with a push-pull power triode circuit. But that is a topic for another article!

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